1.1 TELECOMMUNICATIONS INDUSTRY ASSOCIATION

(TIA)

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[Gray shading indicates required elements]

CONTRIBUTION TO:

TR-41.3.11 Meeting

The Contribution to which this is attached is submitted to a Formulating Group or sub-element thereof of the Telecommunications Industry Association (TIA) in accordance with the provisions of Sections 6.4.1-6.4.6 inclusive of the TIA Engineering Manual, all of which provisions are hereby incorporated by reference.

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ABSTRACT:

This contribution was developed by starting with TIA-470.110-C. The transmission performance section was then revised based on IEEE-269. (The acoustic stability section still needs to be replaced with a TCLw requirement.) The references, abbreviations, acronyms, and definitions sections along with the annexes still need to be revised. The main goal of this contribution is to provide a platform for discussion of the high level organization of the standard and to identify some of the basic philosophical and practical choices that need to be made by the Working Group early in the project.

It is proposed that this contribution be accepted by the TR41.3.11 Working Group as Draft 01 of SP-3-4350.130-RV3 (to be published as TIA-470.130-C).

To be ANSI/TIA-470.130-C

Telecommunications

Telephone Terminal Equipment

Headset Acoustic Performance Requirements for Analog Telephones

Formulated under the cognizance of TIA Subcommittee TR-41.3,
Analog and Digital Wireline Terminals

With the approval of TIA Engineering Committee TR-41,
User Premises Telecommunications Requirements

The TR-41.3.11 Working Group acknowledges the contributions made by the following individuals in the development of this standard.

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The TR-41.3.11 Working Group, which had the technical responsibility during the development of this standard, had the following participating members.

Name	Representing	

Name	Representing	

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2 SCOPE

This standard provides headset acoustic performance requirements for Customer Premises Equipment (CPE) intended for connection to the Public Switched Telephone Network (PSTN). These requirements should ensure compatibility and satisfactory performance to the user in a high percentage of installations. Test measurement methods reference procedures in IEEE Std 269 where applicable.

Several performance measurement procedures are established today, each of which yields standardized measurement data that may be used for the comparison of different products. This document may reference only a single procedure or specific equipment, however the intent is not to be all-inclusive. Any measurement procedure and equipment that can result in an identical measurement is considered valid.

While the procedures may call out specific test points within the requirements, the full range of the requirements take precedent.

3 REFERENCES

The following standards contain provisions, which, through reference in this text, constitute provisions of this standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below. ANSI and TIA maintain registers of currently valid national standards.

3.1 NORMATIVE REFERENCES

1. **ANSI/IEEE Std 269-2002** – *Standard Methods for Measuring Transmission Performance of Analog and Digital Telephone Sets, Handsets and Headsets*
2. **ANSI S1.4-1983 (Reaff. 2001)** – *Specification for Sound Level Meters*
3. **47 CFR Part 68** – *Code of Federal Regulations (CFR), Title 47, FCC Part 68, Connection of Terminal Equipment to the Telephone Network.*
4. **IEEE Std 743-1995** – *IEEE Standard Equipment Requirements and Measurement Techniques for Analog Transmission Parameters for Telecommunications*
5. **ITU-T Recommendation O.41 (94)** –*Psophometer for use on telephone-type circuits*
6. **ITU-T Recommendation P.57 (2002)** – *Artificial Ears*
7. **ITU-T Recommendation P.64 (1999)** – *Determination of Sensitivity/Frequency Characteristics of Local Telephone Systems.*
8. **ITU-T Recommendation P.79 (1999)** – *Calculation of Loudness Ratings for Telephone Sets*
9. **TIA/EIA-504** – *Magnetic Field Intensity Criteria for Telephone Compatibility with Hearing Aids.*

3.2 INFORMATIVE REFERENCES

1. **ANSI/T1.508-2004** – *Loss Plan for Evolving Digital Networks.*
2. **ANSI/TIA-464-C-2002** – *Requirements for PBX Switching Equipment.*
3. **ANSI/TIA-464-C-1-2004** – *Requirements for PBX Switching Equipment, Addendum.*
4. **ANSI/TIA/EIA-810-A-2000** – *Transmission Requirements for Narrowband Voice over IP and Voice over PCM Digital Wireline Telephones.*
5. **ANSI/TIA-912-A-2004** – *Voice Gateway Transmission Requirements*.
6. **ANSI/TIA-968-A-2002** – *Telecommunications Telephone Terminal Equipment Technical Requirements for Connection of Terminal Equipment to the Telephone Network*
7. **ITU-T Recommendation G.100.1 (2001)** – *The use of the decibel and of relative levels in speechband telecommunications.*

8. **ITU-T Recommendation P.58 (1996)** – *Head and Torso Simulator for Telephonometry*
9. **TIA-920-2002** – *Transmission Requirements for Wideband Digital Wireline Telephones.*
10. **UL 60950-1 Issue 3 (2003)** - *Information Technology Equipment - Safety - Part 1: General Requirements*

4 ABBREVIATIONS, ACRONYMS, AND DEFINITIONS

4.1 ABBREVIATIONS AND ACRONYMS

For the purposes of this standard, the following abbreviations and acronyms apply.

ANTHD Amplitude Normalized Total Harmonic Distortion

AWG American Wire Gauge

CPE Customer Premises Equipment

DRP Drum Reference Point

ECRP Ear-Cap Reference Point

ERP Ear Reference Point

HATS Head And Torso Simulator

MRP Mouth Reference Point

OLR Objective Loudness Rating

ONS On premises station

OPS Off premises station

PSTN Public Switched Telephone Network

PBX Private Branch Exchange

RCJ Reference Corner Jig

RETP Receive Electrical Test Point

RLR Receive Loudness Rating

ROLR Receive Objective Loudness Rating

RTP Recommended Test Position

SETP Send Electrical Test Point

SLR Send Loudness Rating

SOLR Send Objective Loudness Rating

STMR Sidetone Masking Rating

T&R Tip and Ring terminals or interface

THD Total Harmonic Distortion

TOLR Transmit Objective Loudness Rating

4.2 DEFINITIONS

For the purposes of this standard, the following definitions apply.

+12 dB Volume Control Setting: The setting at which the measured RLR with 0 km, 2.7 km, and 4.6 km artificial lines, is closest to, but not less than, 12 dB louder than the RLR at the Reference Volume Control Setting.

Amplitude Normalized Total Harmonic Distortion (ANTHD): The ratio of the square root of the sum of all of the squared second, third, and higher harmonic amplitudes, at their harmonic frequencies, (normalized to the amplitude of the fundamental at the same frequencies) to the amplitude of the fundamental.

For this document ANTHD is calculated as a percentage using the second and third harmonics. The formula is:

$$\%\ ANTHD \quad = \quad 100 \sqrt{\left[\frac{H_2(f)}{H_1(2f)}\right]^2 + \left[\frac{H_3(f)}{H_1(3f)}\right]^2}$$

where:

$H_1(2f)$ is the amplitude of the fundamental at frequency 2f.

$H_1(3f)$ is the amplitude of the fundamental at frequency 3f.

$H_2(f)$ is the amplitude of the second harmonic at excitation frequency f.

$H_3(f)$ is the amplitude of the third harmonic at excitation frequency f.

Central Office: Used broadly to describe the switch connected to a subscriber's line.

CPE State: The condition of the telephone device. It can be either On-Hook or Off-Hook.

dBA: Sound pressure level in decibels, relative to 2×10^{-5} Pa, using A-weighting. (See Annex A)

dBm: Power level in decibels, relative to a power of 1 mW (milliwatt).

dBmp Power level in decibels, relative to a power of 1 mW, using psophometric weighting. (See Annex A)

dBPa: Sound pressure level in decibels, relative to a sound pressure of 1 Pa (pascal).

dBV: Voltage level in decibels, relative to 1 volt rms.

End Office: A central office to which an end user telephone is connected.

HATS Position: The handset positioning specified by Annex D or E of ITU-T Recommendation P.64.

Listening-Path: The path from electrical signals at the RETP to acoustic signals at the CPE receiver/speaker.

Mouth Reference Point (MRP): A point on the axis of the mouth simulator, 25 mm in front of the center of the equivalent lip plane.

Off-Hook: The state of the CPE when the hook-switch is closed. Refers to the state of a particular CPE rather than the line state.

On-Hook: The state of the CPE when the hook-switch is open. Refers to the state of a particular CPE rather than the line state.

Off-hook noise: The internally generated noise of the telephone set. It is the psophometrically weighted signal power delivered to a specified termination (SETP) in the absence of an acoustic input into the telephone set.

Reference Volume Control Setting: The quietest volume control setting that complies with the mandatory low leak RLR requirement at 0 km, 2.7 km, and 4.6 km artificial lines.

Note: For many telephones, the Reference Volume Control Setting on the Type 3.3 and Type 3.4 ears may be a louder setting than on the Type 1 ear, which was specified in previous revisions of this standard, due to the acoustic leakage on the Type 3.3 and Type 3.4 ears.

Talking-Path: The path from acoustic signals at the CPE microphone to electrical signals at the SETP.

Nominal LR The loudness rating used for network loss planning.

Telephone set: A device that facilitates two-way voice communication between persons over the public switched telephone network. A telephone set consists of:

- A transmitter and a receiver that perform the conversions between acoustical energy and electrical energy.
- A speech network that couples the transmitter and receiver circuits to the telephone line.
- A hookswitch that opens and closes electrical contacts to indicate the status (off-hook or on-hook) of the telephone set to the central office switch.
- A network-addressing unit, which generates dial pulses or DTMF signals.
- An alerting device that produces a signal to indicate an incoming call attempt on the telephone line.

Total Harmonic Distortion (THD): The ratio of the square root of the sum of all of the squared second, third, and higher harmonics to the magnitude of the fundamental.

For this document THD is calculated as a percentage using the second and third harmonics. The formula is:

$$\%\,THD \;=\; 100\,\frac{\sqrt{[H_2(f)]^2 + [H_3(f)]^2}}{H_1(f)}$$

where:

$H_1(f)$ is the amplitude of the fundamental at frequency f.

$H_2(f)$ is the amplitude of the second harmonic at excitation frequency f.

$H_3(f)$ is the amplitude of the third harmonic at excitation frequency f.

5 TECHNICAL REQUIREMENTS

5.1 CATEGORIES OF CRITERIA

Four types of requirements are specified in this standard: Mandatory, Recommended, Permissive and Advisory:

1. **Mandatory requirements** are designated by the terms "shall" and "shall not." These requirements are used to indicate conformity in which no deviation is permitted.
2. **Recommended requirements** are designated by the terms "should" and "should not." These requirements generally relate to compatibility or performance advantages towards which future designs should strive.
3. **Permissive requirements** are designated by the terms "may" and "may not." These requirements are used to indicate an action that is permitted within the limits of the standard.
4. **Advisory requirements** are designated by the term "desirable." Advisory criteria represent product goals or are included in an effort to ensure universal product compatibility and may be used instead of a Recommended requirement. If both a mandatory and an advisory level are specified for the same criterion, the advisory level represents a goal currently identifiable as having distinct compatibility or performance advantages toward which future designs should strive.

5.2 GENERAL

The transmission performance of an analog telephone with headset is determined by its acoustical-to-electrical, electrical-to-acoustical, or acoustical-to-acoustical transfer characteristics.

The default test signal in this document is sinusoidal with the R40 format preferred (see Annex C). Some telephone designs may include non-linear processes (which may not respond reliably to sinusoidal signals); thus it may be advisable to use artificial speech as a test signal as described in IEEE Std 269.

Mouth and ear simulator selection, calibration, usage and headset positioning shall be done according to IEEE Std 269 unless stated otherwise in this document.

A Type 3.3 or a Type 3.4 ear simulator (see ITU-T Recommendation P.57) shall be used for all measurements. Although the 3.3 and 3.4 artificial ears are similar, they are not identical. Therefore, the same type of artificial ear shall be used for all tests and the ear type shall be documented.

5.2.1 Recommended Test Position

If a Recommended Test Position (RTP) has been specified by the headset manufacturer, then the headset shall be tested in that position.

5.2.2 Estimated Real Use Position

If the headset to be tested does not have a manufacturer specified RTP, then an Estimated Real Use Position (ERUP) shall be used. The ERUP shall be determined by following the guidelines in IEEE-269 clause 5.3.3 and the following sub-clauses.

5.2.2.1 Headset earphone positioning

IEEE-269-2002 and ITU-T P.380 Preliminary Draft state that headset positioning is to be defined geometrically with respect to the MRP, center of the lip plane, ERP, ear entrance point (EEP) and/or the HATS reference point (HRP). These documents do not permit the use of facial features (e.g. the corner of the mouth) as reference points. However, in practice it can be difficult to precisely position the headset earphone using only geometric terms relative to the specified reference points. This difficulty is due to the fact that the Ear Reference Point (ERP) and the Ear canal Entrance Point (EEP) are not visible after the earphone is initially placed over the pinna. Also, there are no headset positioning jigs currently provided by the HATS manufacturers. Therefore, additional documentation regarding earphone placement using descriptive wording and/or photographs is desirable.

In this document, confidence in the accuracy of earphone placement is enhanced by performing the receive response measurement five times and then calculating the average and standard deviation of the results (as recommended by IEEE-269 and P.380).

5.2.2.2 Headset microphone positioning

The ERUP headset microphone position shall be defined in three coordinates based on the HATS Center of Lips (CL) as the origin (per ITU-T P.64-1999 Annex E). The X_m axis (on mouth axis displacement) coincides with the mouth reference axis and has positive direction into the mouth. The Y_m axis (off mouth axis horizontal displacement) is horizontal, perpendicular to the X_m axis with positive direction towards the right side of the head. The Z_m axis (off mouth axis vertical displacement) is perpendicular to the X_m axis and Y_m axis with positive direction upwards.

For example, the ERUP microphone position could be defined as X_m = - 6 mm, Y_m = +42 mm, and Z_m = -9 mm. This would mean that the microphone is located 6mm in front, 42mm to the right, and 9mm below the HATS Center of Lips (CL).

Note: Pressure gradient microphones (cardioid, noise canceling etc.) are sensitive to both position and orientation. It is important that the correct orientation be used to obtain results representing actual use performance.

5.3 TEST CIRCUITS

The following test circuits are referenced in this document:

5.3.1 Send Testing with Artificial Lines

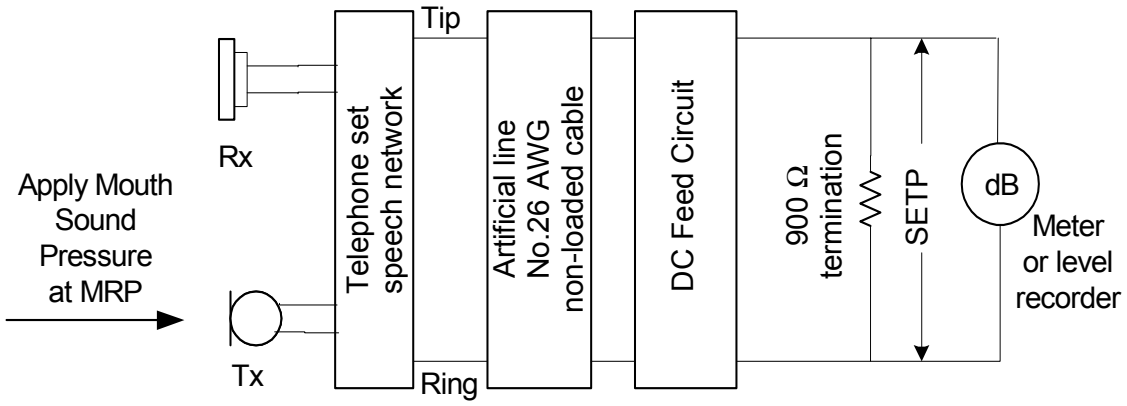


Figure 1 – Test circuit for send measurements

5.3.2 Receive Testing with Artificial Lines

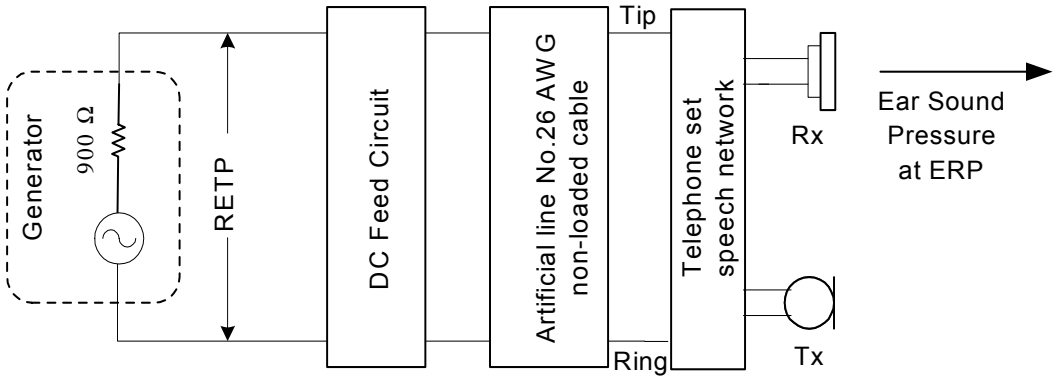


Figure 2 – Test circuit for receive measurements

5.3.3 Sidetone Testing with Artificial Lines

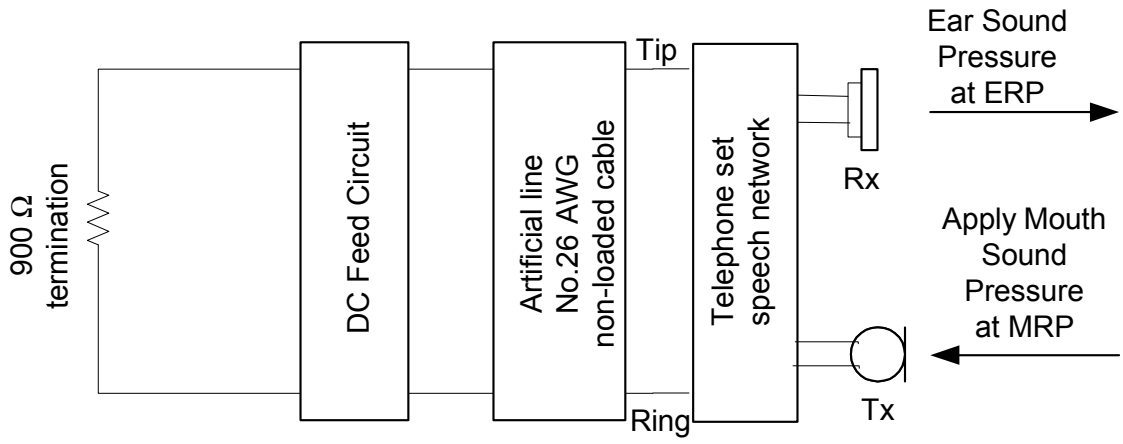


Figure 3 – Test circuit for sidetone measurements

5.3.4 DC Feed Circuit

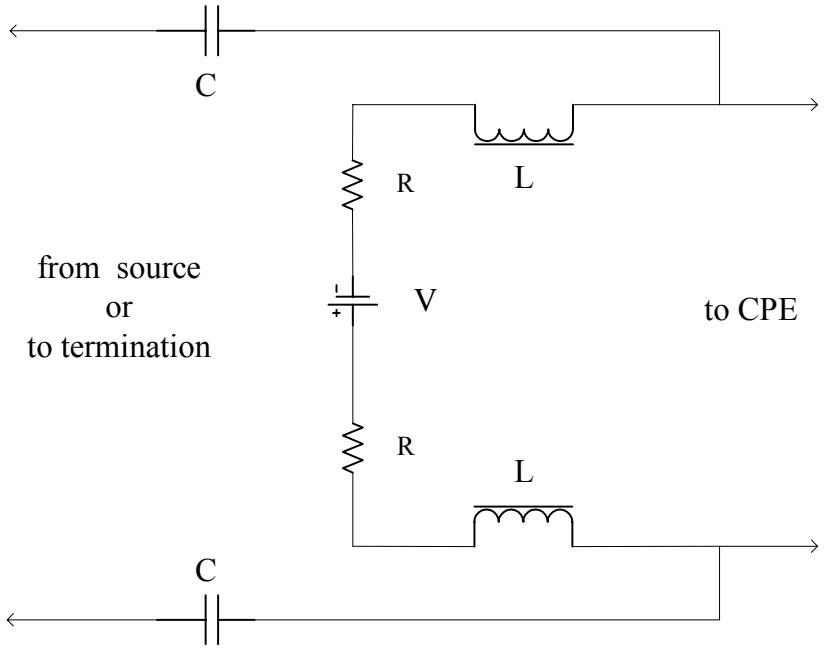
The dc feed circuit must supply the CPE power and interface to test equipment such that the required parameters may be measured without introducing any significant error. The dc feed shall provide the following:

- 50 Vdc in series with 400 ohms +/- 1%
- less than 0.1 dB insertion loss over the 100 Hz to 10 kHz range

NOTE: See IEEE Std 269 for the procedure to follow when determining feed circuit loss.

- At least 30 dB return loss with respect to 900 ohms over the 300 Hz to 10 kHz range

One possible implementation of the dc Feed is shown in Figure 4.



NOTES:

1. C >= 100 uF with pair having matched values within 1%.
2. L >= 5 H with pair having matched values within 20%.
3. R = 200 ohms (including resistance of L) with pair having matched values within 1%.
4. V = 50 Vdc +/- 2 Vdc.
5. This is a simplified DC feed circuit. It does not implement the current limit characteristic of modern end offices, which typically limit the maximum current supplied by the office.

Figure 4 – One possible dc feed circuit

5.3.5 Termination Impedance

The termination impedance shall be a resistive load of 900 ohms with a tolerance of +/- 1 %.

5.3.6 Artificial Lines

This standard has transmission requirements for CPE when connected to 0 km, 2.7 km and 4.6 km artificial lines. These lines may be assembled using combinations of artificial line segments shown in Figure 5 with component values selected from Table 1. These artificial line segments represent 26 AWG non-loaded cable.

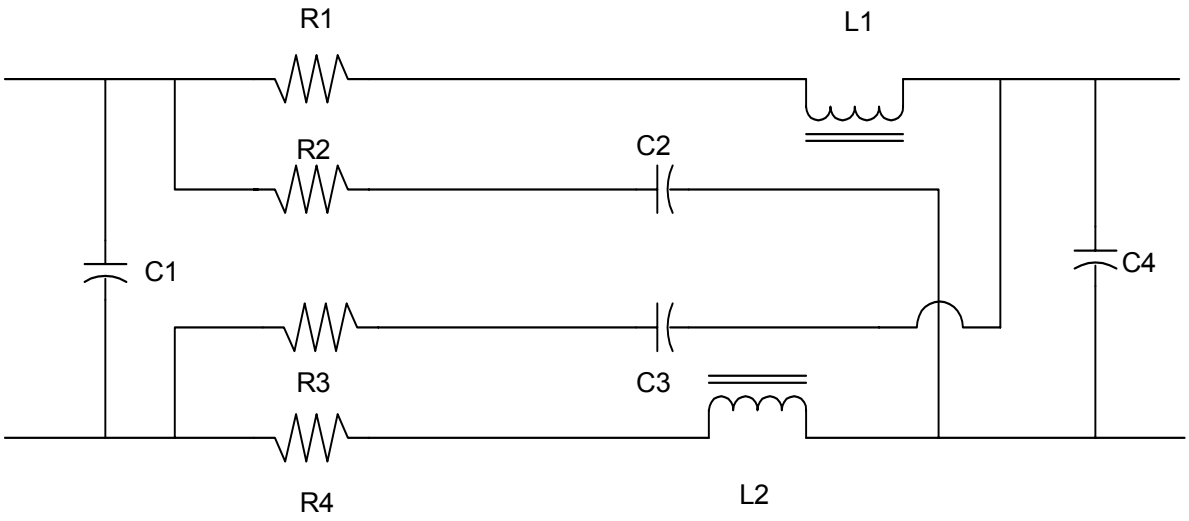


Figure 5 – Artificial line segment

Table 1 – Artificial line segment component values

Component	0.305 km (1 kft)	0.914 km (3 kft)	1.83 km (6 kft)
R_1, R_4	41.7 Ω	124 Ω	249 Ω
R_2, R_3	109 Ω	174 Ω	312 Ω
C_1, C_4	3.77 nF	11.3 nF	22.6 nF
C_2, C_3	4.02 nF	12.2 nF	25.5 nF
L_1, L_2	96.0 µH	336 µH	983 µH

NOTES:
(1) *All values are ± 1 %.*
(2) 2.7 km *(9 kft) and* 4.6 km *(15 kft) can be made up of cascaded sections of the above.*

5.3.7 CPE Load Simulator

The CPE Load Simulator shall model the dc V-I characteristic in Figure 6 (as measured at T&R, within ± 0.3 V). The CPE Load Simulator shall also provide a 600 ohm impedance and have a return loss of greater than 10 dB with respect to 600 ohms over the 200 Hz to 4000 Hz range at the T&R terminals. The Return Loss shall meet this requirement for dc currents between 0 and 60 mA.

One possible implementation of the CPE Load Simulator is shown in Figure 7.

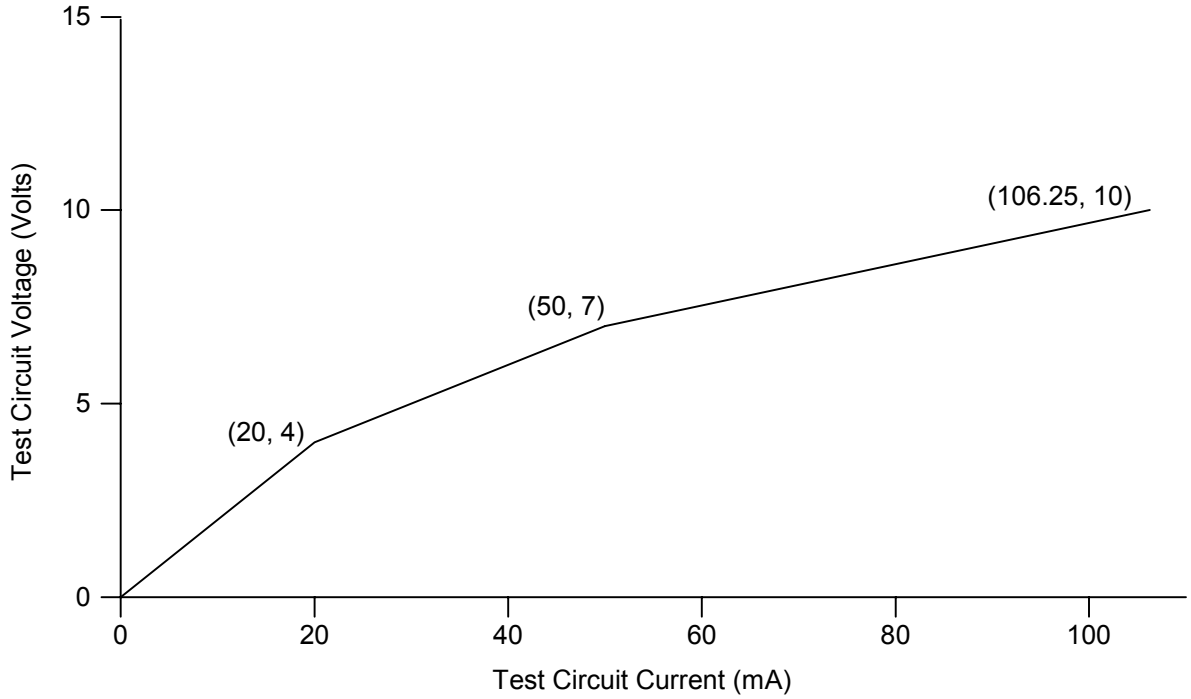
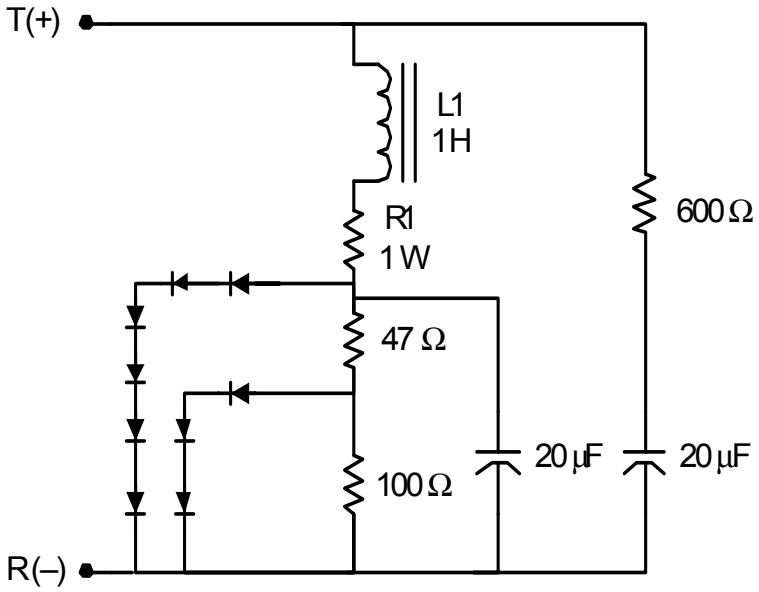


Figure 6 – CPE load simulator V-I characteristic requirement



NOTES:

1. Circuit is polarity sensitive.
2. Use IN4004, or similar, for diode strings.
3. L1 may be > 1 H
4. Resistor values shall be ± 1 %.
5. R1 + L1 coil resistance should be 53 ohms.
6. The performance of this circuit is dependent on the characteristics of the diodes used and temperature. Performance of the circuit must be verified and component values may need to be adjusted to meet the requirements.

Figure 7 – One possible CPE load simulator implementation

5.4 SEND PERFORMANCE

5.4.1 General

The send performance of an analog telephone with headset is determined by its acoustical-to-electrical transfer characteristics. To determine the send performance, requirements for frequency response, Send Loudness Rating (SLR), linearity, distortion, noise and, if supported, muting are specified.

If the headset is equipped with a send gain adjustment, the gain control shall be set to the manufacturer's default setting. If no default setting is defined by the manufacturer then the send gain control shall be set to the middle of its adjustable range.

5.4.2 Headset Send Frequency Response

5.4.2.1 Requirement

The send frequency response shall be measured using 0 km, 2.7 km, and 4.6 km artificial lines. The send frequency response recorded for a 2.7 km artificial line shall fall within the upper and lower limits in Table 2 (shown in Figure 8) between 100 Hz and 8 kHz. These are floating limits, which are positioned vertically for the best fit to the actual measurement.

Table 2 – Headset send frequency response limits

Headset send response limits		
Frequency (Hz)	**Lower Limit (dB)**	**Upper Limit (dB)**
100		-4
200		2
300	-12.5	
1000	-2	2
2000	-4	12
3000	-12.5	
4000		10
8000		-20

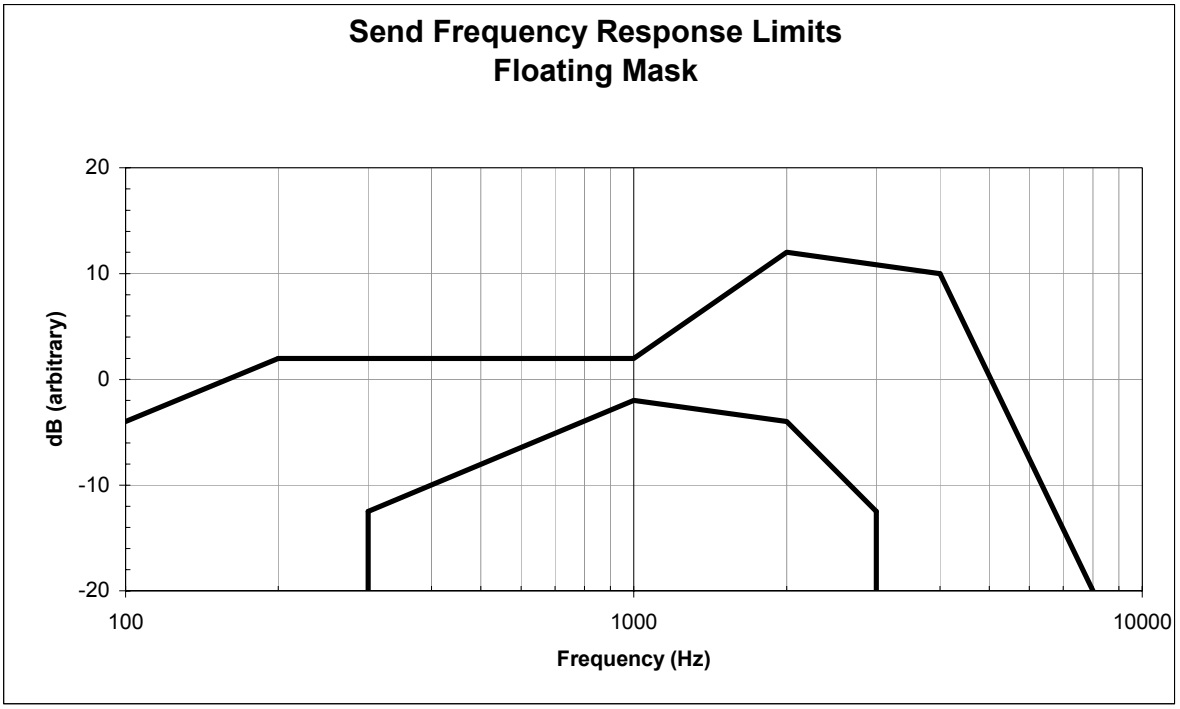


Figure 8 – Headset send frequency response limits

5.4.2.2 Method of Measurement

The frequency response curves shall be measured over the range of 100 Hz to 8 kHz using the test circuit shown in Figure 1 with a signal level of -4.7 dBPa at the MRP.

5.4.3 Headset Send Loudness Rating (SLR)

5.4.3.1 Basic Requirement

The SLR values for 0 km, 2.7 km, and 4.6 km artificial lines shall be calculated from the send frequency response measurements (5.4.2) as described in Annex B and shall fall within the ranges given in Table 3.

Table 3 – Headset send loudness rating limits

Headset Send Loudness Rating (SLR)		
Artificial line, km	**Nominal dB**	**Mandatory range, dB**
0	6	17 to 1
2.7	8	19 to 3
4.6	10	21 to 5

NOTES:

1. Loudness rating values represents system losses. More positive values of SLR represent quieter send levels.
2. The nominal SLR values are based on ITU-T G.101. (See Annex F)
3. See Annex E for more information on Loudness Ratings of telephones on a PBX or a DLC.

5.4.3.2 Parallel Set Off-Hook Requirement

The SLR values measured for determining compliance with 5.4.3.1 shall not change by more than 8 dB for any of the 0 km, 2.7 km and 4.6 km artificial line conditions when the CPE Load Simulator is connected in parallel with the set under test. Note: This requirement only applies to CPE that are designed to operate only with a headset (no support for a handset).

5.4.3.3 Conference Mode

5.4.3.3.1 Requirement

For multi-line sets, the SLR values measured for determining compliance with 5.4.3.1 shall not change by more than 8 dB for any of the 0 km, 2.7 km and 4.6 km artificial line conditions when conferenced to a second line connected to an identical artificial line condition. Note: This requirement only applies to multi-line CPE that are designed to operate only with a headset (no support for a handset).

5.4.3.3.2 Method of Measurement

1. Two lines of the set shall be connected to identical battery feeds and artificial line arrangements as the line tested for compliance with 5.4.3.1.
2. The telephone shall be placed into a mode where the two lines are conferenced together and the SLR measurement shall be repeated on the original line to determine compliance with 5.4.3.3.

5.4.4 Headset Send Linearity

The CPE shall meet either the sinusoidal stimulus requirement or the artificial speech stimulus requirement.

5.4.4.1 Requirement (Sinusoidal Stimulus)

The send linearity response curves for the -20 dB and +10 dB levels with respect to the reference level (-4.7 dBPa) measured in 5.4.2, using a 2.7 km artificial line, shall be within the values listed in Table 4, for the R40 frequencies 300 Hz to 3150 Hz. The Mouth simulator shall have an output distortion less than 2 % THD at +5.3 dBPa.

Table 4 – Send linearity requirements (Sinusoidal)

Level from Reference	Frequency Response Tolerance
-20 dB	+/- 3.0 dB
+10 dB	+/- 2.0 dB

5.4.4.2 Method of Measurement

The following steps shall be taken:

1. Determine the reference linearity response from the 2.7 km artificial line measurement in 5.4.2.
2. Perform a frequency response measurement at the -20 dB relative source level (-24.7 dBPa) and determine the linearity response versus the reference.
3. Perform a frequency response measurement at the +10 dB relative source level (+5.3 dBPa) and determine the linearity response versus the reference.

5.4.4.3 Requirement (Artificial Speech Stimulus)

The send linearity response for the -10 dB and -20 dB levels shall be measured with respect to the Reference level (-4.7 dBPa) measured in 5.4.2, using a 2.7 km artificial line. The tolerance values listed in Table 5 shall be applied for frequencies from 300 Hz to 3150 Hz.

Table 5 – Send linearity requirements (Artificial Speech)

Level from Reference	Frequency Response Tolerance
-20 dB	+/- 3.0 dB
-10 dB	+/- 2.0 dB

5.4.4.4 Method of Measurement

The following steps shall be taken:

1. Determine the reference linearity response from the 2.7 km artificial line measurement in 5.4.2.
2. Perform a frequency response measurement at the -10 dB relative source level (-14.7 dBPa) and determine the linearity response versus the reference.
3. Perform a frequency response measurement at the -20 dB relative source level (-24.7 dBPa) and determine the linearity response versus the reference.

5.4.5 Headset Send Distortion

5.4.5.1 Headset Send Amplitude Normalized Total Harmonic Distortion (ANTHD)

Send distortion has previously been specified as THD, but to eliminate the effects of the send frequency response, ANTHD is used in this document.

The send ANTHD values with 0 km and 4.6 km artificial lines shall be determined at each of the listed frequencies and shall be less than the values listed in Table 6. The Mouth simulator shall have an output distortion less than 2 % THD at +5.3 dBPa.

Table 6 – Headset Send ANTHD requirements.

Frequency [Hz]	Mouth Pressure Level at MRP	
	-4.7 dBPa	+5.3 dBPa
300	5%	n/a
500	5%	10%
800	5%	n/a
1000	5%	10%

5.4.5.2 Headset Send ANTHD with a Parallel set

The send ANTHD values for 0 km and 4.6 km artificial lines shall not exceed 10% for 500 Hz and 1000 Hz, using -4.7 dBPa as the level at the MRP, with the CPE Load Simulator connected in parallel with the set under test. Note: This requirement only applies to CPE that are designed to operate only with a headset (no support for a handset).

5.4.6 Headset Send Noise

The send noise measurements shall be made as described in IEEE Std 269 using the psophometric method only.

5.4.6.1 Headset Off-Hook Noise

5.4.6.1.1 Requirement

The psophometrically weighted telephone set noise (see Annex A) shall not exceed -69 dBmp when measured with 0 km, 2.7 km and 4.6 km artificial lines over a minimum period of 5 seconds.

5.4.6.1.2 Method of Measurement

The telephone set noise shall be measured using the test circuit shown in Figure 1 except no signal shall be applied at the MRP. The noise shall be measured under the following conditions:

1. The measurements shall be made in the Off-Hook mode with 0 km, 2.7 km, and 4.6 km artificial lines.
2. The microphone shall be isolated from sound input and mechanical disturbances. The background noise level shall be less than 29 dBA (as specified in IEEE Std 269).
3. The measurement shall be taken over a minimum period of 5 seconds.
4. Microphone Muting shall not be used to provide room noise isolation.

5.4.6.2 Headset On-Hook Noise

5.4.6.2.1 Requirement

The telephone set noise shall not exceed -80 dBmp with a 0 km artificial line measured over a minimum period of 5 seconds. Note: This requirement only applies to CPE that are designed to operate only with a headset (no support for a handset).

5.4.6.2.2 Method of Measurement

The telephone set noise shall be measured using the test circuit shown in Figure 1 except no signal shall be applied at the MRP.

5.4.6.3 Headset Off-Hook Send Single Frequency Interference

5.4.6.3.1 General

Narrow-Band noise, including single frequency interference, is an impairment that can be perceived as a tone depending on its level relative to the overall weighted noise level. This test measures the weighted noise level characteristics in narrow bands of not more than 31 Hz.

5.4.6.3.2 Requirement

The send single frequency interference shall not exceed -78 dBmp with 0 km, 2.7 km and 4.6 km artificial lines when measured over a minimum period of 5 seconds.

5.4.6.3.3 Measurement Method

The telephone set noise shall be measured using the test circuit shown in Figure 1 except no signal shall be applied at the MRP. The noise shall be measured under the following conditions:

1. The measurements shall be made in the Off-Hook mode with 0 km, 2.7 km, and 4.6 km artificial lines.
2. The microphone shall be isolated from sound input and mechanical disturbances. The background noise level shall be less than 29 dBA (as specified in IEEE Std 269).
3. Measure the psophometrically-weighted noise level with a selective voltmeter or spectrum analyzer with an effective bandwidth of not more than 31 Hz, over the frequency range of 100 to 3500 Hz.
4. If FFT analysis is used, then “Flat Top” windowing shall be employed.
5. Microphone Muting shall not be used to provide room noise isolation.

5.4.7 Headset Send Muting

5.4.7.1 Requirement

If supported, the send muting shall be determined over the frequency range of 100 Hz to 8000 Hz. During muting, any transmitted signal in the frequency range for 0 km, 2.7 km, and 4.6 km artificial lines shall not exceed -80 dBV at the SETP.

5.4.7.2 Method of Measurement

The send muting shall be measured using the test circuit shown in Figure 1.

1. Set the Mouth simulator level to +5.3 dBPa.
2. The CPE Muting measurement shall made be with 0 km, 2.7 km and 4.6 km artificial lines.
3. This measurement should be done with a narrow band meter that has a noise floor less than –86 dBV.

5.5 RECEIVE PERFORMANCE

5.5.1 General

The receive performance of an analog telephone with headset is determined by its electrical-to-acoustical transfer characteristics. To determine the receive performance, requirements for frequency response, Receive Loudness Rating (RLR), linearity, distortion, noise and volume control are specified.

The CPE's receive volume control shall be set to the Reference Volume Control Setting when checking for compliance with the requirements for receive characteristics, unless specifically stated otherwise. The stimulus levels specified for receive tests are specified as 900 ohm nominally loaded (one-half of the open circuit voltage). Following a calibration, the resistive load is removed and the source is connected to the CPE without further adjustment.

5.5.2 Headset Receive Frequency Response

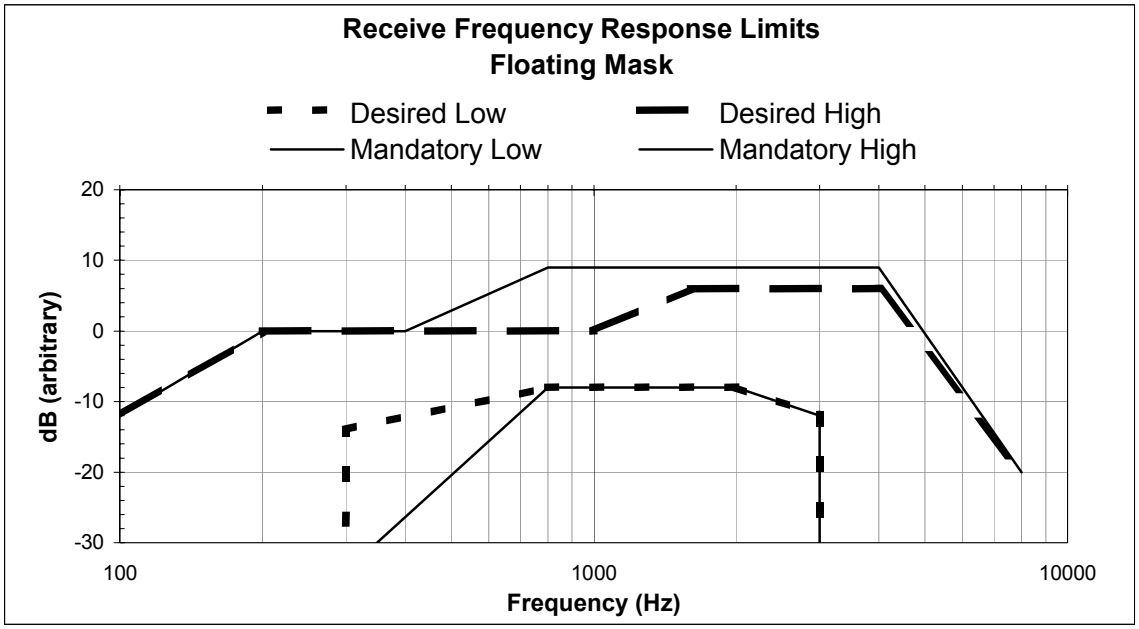
5.5.2.1 Requirement

The receive frequency response shall be measured using 0 km, 2.7 km, and 4.6 km artificial lines.

The mean receive frequency response between 100 Hz and 8 kHz (referenced to the ERP, see IEEE Std 269) for a 2.7 km artificial line shall fall within the mandatory limits in Table 7 (shown in Figure 9) and should fall within the desired limits. These are floating limits, which are positioned vertically for the best fit to the actual measurement.

Table 7 – Headset receive frequency response limits

Headset receive response limits				
Frequency (Hz)	***Desired Lower Limit (dB)***	**Mandatory Lower Limit (dB)**	**Mandatory Upper Limit (dB)**	***Desired Upper Limit (dB)***
100			-12	*-12*
200			0	*0*
300	*-14*	-34		
400			0	
500				
800	***-8***	-8	9	
1000				*0*
1600				*6*
2000	*-8*	-8		
3000	*-12*	-12		
4000			9	*6*
8000			-20	*-20*



527

528 **Figure 9 – Headset receive frequency response limits**

529

530 **5.5.2.2 Method of Measurement**

531 The frequency response curves shall be measured over the range of 100 Hz to 8 kHz using the test
532 circuit shown in Figure 2 with the generator level set for a nominally loaded level of -16 dBV. For
533 binaural headsets, after correction to the ERP, the measured receive frequency responses for the right
534 and left ears shall be summed on a voltage basis. The headset shall then be completely removed from
535 the HATS and then re-mounted on the HATS. This process shall be repeated until a total of five
536 measurements have been made. The mean and standard deviation of each point of the frequency
537 response shall be computed from this data. The measurement shall be considered valid if the
538 standard deviation of the frequency response is 2dB or less from 200 to 1000 Hz and 1dB or less
539 from 1000 to 4000 Hz.

540

5.5.3 Headset Receive Loudness Rating (RLR)

5.5.3.1 Basic Requirement

The RLR values for 0 km, 2.7 km, and 4.6 km artificial lines shall be calculated from each of the five receive frequency response measurements (5.5.2) as described in Annex B. The measurement shall be considered valid if the standard deviation of the loudness rating is 1dB or less. The mean RLR shall fall within the mandatory range given in Table 8 and should fall within the desired range.

Table 8 – Headset receive loudness rating limits

Headset Receive Loudness Rating (RLR)			
Artificial Line, km	**Nominal, dB**	**Mandatory Range, dB**	**Desired Range, dB**
0	–5	+3 to –10	0 to –10
2.7	–3	+5 to –8	+2 to –8
4.6	–1	+7 to –6	+4 to –6

NOTES:

1. Loudness rating values represents system losses. More positive values of RLR represent quieter receive levels.
2. See Annex E for more information on Loudness Ratings of telephones on a PBX or a DLC.

5.5.3.2 Parallel Set Off-Hook Requirement

There shall be no more than 8 dB difference between the RLR values measured for determining compliance with 5.5.3.1 and the RLR with any of the specified artificial line conditions and the CPE Load Simulator connected in parallel with the set under test. Note: This requirement only applies to CPE that are designed to operate only with a headset (no support for a handset).

5.5.3.3 Conference Mode Requirement

5.5.3.3.1 Requirement

For multi-line sets, the RLR values measured for determining compliance with 5.5.3.1 shall not change by more than 8 dB for any of the 0 km, 2.7 km and 4.6 km artificial line conditions when conferenced to a second line connected to an identical artificial line condition. Note: This requirement only applies to multi-line CPE that are designed to operate only with a headset (no support for a handset).

5.5.3.3.2 Method of Measurement

1. Two lines of the set shall be connected to identical battery feeds and artificial line arrangements as the line tested for compliance with 5.5.3.1.
2. The telephone shall be placed into a mode where the two lines are conferenced together and the RLR measurement shall be repeated on the original line to determine compliance with 5.5.3.3.

5.5.4 Headset Receive Linearity

The CPE should meet either the sinusoidal requirement or the artificial speech requirement.

5.5.4.1 Requirement (Sinusoidal Stimulus)

The receive linearity response curves for the -20 dB and +10 dB levels with respect to the reference level (-16 dBV) measured in 5.5.2, using a 2.7 km artificial line, shall be within the values listed in Table 9, for the R40 frequencies 300 Hz to 3150 Hz.

Table 9 – Headset receive linearity requirements (Sinusoidal)

Level from Reference	Frequency Response Tolerance
-20 dB	+/- 3.0 dB
+10 dB	+/- 2.0 dB

5.5.4.2 Method of Measurement

The following steps shall be taken:

1. Determine the reference linearity response from the 2.7 km artificial line measurement in 5.5.2.
2. Perform a frequency response measurement at the -20 dB relative source level (-36 dBV) and determine the linearity response vs. the reference.
3. Perform a frequency response measurement at the +10 dB relative source level (-6 dBV) and determine the linearity response vs. the reference.

5.5.4.3 Requirement (Artificial Speech Stimulus)

The receive linearity response for the -20 dB and -30 dB levels shall be measured with respect to the reference level (-16 dBV) measured in 5.5.2, using a 2.7 km artificial line. The tolerance values listed in Table 10 shall be applied for frequencies from 300 Hz to 3150 Hz.

Table 10 – Headset receive linearity requirements (Artificial Speech)

Level from Reference	Frequency Response Tolerance
-30 dB	+/- 4.0 dB
-20 dB	+/- 3.0 dB

5.5.4.4 Method of Measurement

The following steps shall be taken:

1. Determine the reference linearity response from the 2.7 km artificial line measurement in 5.5.2.
2. Perform a frequency response measurement at the -20 dB relative source level (-36 dBV) and determine the linearity response versus the reference.
3. Perform a frequency response measurement at the -30 dB relative source level (-46 dBV) and determine the linearity response versus the reference.

5.5.5 Headset Receive Distortion

5.5.5.1 Headset Receive Amplitude Normalized Total Harmonic Distortion (ANTHD)

Receive distortion has previously been specified as THD, but to eliminate the effects of the receive frequency response, ANTHD is used in this document.

The receive volume control shall be set at the reference setting. For the increased receive volume measurement, the control shall be set to its +12 dB Volume Control Setting. The THD of the signal source shall be less than 0.5% for signals of -16 dBV and shall be less than 1.0% for signals of -1 dBV.

The receive ANTHD values with 0 km and 4.6 km artificial lines shall be determined at each of the listed frequencies and shall be less than the values listed in Table 11.

Table 11 – Headset Receive ANTHD requirements

Frequency [Hz]	RETP Input Level		
	-16 dBV at the Reference Volume Control Setting	**-16 dBV at the +12 dB volume control setting**	**-1 dBV at the Reference Volume Control Setting**
300	5%	10%	n/a
500	5%	10%	10%
800	5%	10%	n/a
1000	5%	10%	10%

5.5.5.2 Receive ANTHD with a Parallel set

At the Reference Volume Control Setting the receive ANTHD values for 0 km, and 4.6 km artificial lines shall not exceed 10% for 500 Hz and 1000 Hz, using –16 dBV as the nominally loaded generator level, with the CPE Load Simulator connected in parallel with the set under test. Note: This requirement only applies to CPE that are designed to operate only with a headset (no support for a handset).

5.5.6 Headset Receive Noise

The receive noise measurements shall be referenced to the ERP.

5.5.6.1 Headset Off-Hook Noise

5.5.6.1.1 Requirement

The acoustic noise level from the telephone handset receiver at the Reference Volume Control Setting shall not exceed 40 dBA for 0 km, 2.7 km, and 4.6 km artificial lines when measured over a minimum period of 5 seconds.

5.5.6.1.2 Method of Measurement

The measurement shall be made as described in IEEE Std 269. The telephone set noise shall be measured using the test circuit shown in Figure 2 except no signal shall be applied at the RETP. The noise shall be measured under the following conditions:

1. The measurements shall be made in the Off-Hook mode with 0 km, 2.7 km, and 4.6 km artificial lines.
2. The microphone shall be isolated from sound input and mechanical disturbances. The background noise level shall be less than 29 dBA (as specified in IEEE Std 269).
3. The measurement shall be taken over a minimum period of 5 seconds.
4. Microphone Muting shall not be used to provide room noise isolation.

5.5.6.2 Headset Off-Hook Receive Single Frequency Interference

5.5.6.2.1 General

Narrow-Band noise, including single frequency interference, is an impairment that can be perceived as a tone depending on its level relative to the overall weighted noise level. This test measures the weighted noise level characteristics in narrow bands of not more than 31 Hz, which can then be compared to the overall weighted background noise level.

5.5.6.2.2 Requirement

The receive single frequency interference at the Reference Volume Control Setting shall be 10 dB quieter than the A-weighted broadband noise floor when measured over a minimum period of 5 seconds.

5.5.6.2.3 Measurement Method

The measurement shall be made as described in IEEE Std 269. The telephone set noise shall be measured using the test circuit shown in Figure 2 except no signal shall be applied at the RETP. The noise shall be measured under the following conditions:

1. The measurements shall be made in the Off-Hook mode with 0 km, 2.7 km, and 4.6 km artificial lines.
2. The microphone shall be isolated from sound input and mechanical disturbances. The background noise level shall be less than 29 dBA (as specified in IEEE Std 269).
3. Microphone Muting shall not be used to provide room noise isolation.

673
674 **5.5.7 Headset Receiver Volume Controls**
675 The receiver volume control performance requirements are below.
676 **5.5.7.1 Requirement**
677 The RLR at the maximum volume control setting shall be at least 12 dB louder than the RLR at the
678 Reference Volume Control Setting (measured in 5.5.3.1) for 0 km, 2.7 km and 4.6 km artificial lines.
679 If the maximum volume control setting RLR is more than 18 dB louder than the RLR at the
680 Reference Volume Control Setting, then the CPE shall automatically reset to the Reference Volume
681 Control Setting after going On-Hook to end the call.
682
683 **5.5.7.2 Method of Measurement**
684 The receive frequency response shall be measured according to 5.5.2.2 with the volume control at the
685 maximum volume control setting using 0 km, 2.7 km, and 4.6 km artificial lines. The RLR shall be
686 calculated from these responses as described in Annex B.
687
688
689

5.6 SIDETONE PERFORMANCE

5.6.1 General

The sidetone performance of an analog telephone with headset is determined by its acoustical-to-acoustical transfer characteristics. To determine the sidetone performance, requirements for Sidetone Masking Rating (STMR) and Sidetone Echo Delay are specified.

The CPE's receive volume control shall be set to the reference setting when checking for compliance with the requirements for sidetone characteristics, unless specifically stated otherwise.

5.6.2 Headset Sidetone Masking Rating (STMR)

5.6.2.1 Requirement

A telephone set shall have Sidetone Masking Rating (STMR) values between 24 and 8 dB for 0 km, 2.7 km and 4.6 km artificial lines.

NOTE: More positive values of STMR represent quieter sidetone levels.

5.6.2.2 Method of Measurement

1. The sidetone frequency response shall be measured over the range of 100 Hz to 8 kHz using the test circuit shown in Figure 3 and with a signal level of –4.7 dBPa at the MRP.
2. The STMR values shall be calculated from these responses as described in Annex B.

5.6.3 Headset Sidetone Echo Delay

Sidetone echo occurs when significant delay is introduced in the speech path between the headset microphone, the 2-4 wire hybrid, and the headset receiver to make the signal distinguishable from normal sidetone. Both normal sidetone and sidetone echo are a portion of the transmitted speech power that is coupled to the receiver of the same telephone set, with the difference being the amount of delay introduced in the speech path. Normal sidetone delay is typically less than 5 ms while sidetone echo delay may be as much as 28 ms. In general, customer satisfaction is decreased as the loss of the echo path is decreased or the delay of the echo path is increased.

5.6.3.1 Recommendation

For sidetone echo delay greater than 8 ms, it is recommended that the CPE incorporate some form of echo control and customer acceptability should be verified subjectively.

NOTE: This issue is under further study.

5.6.3.2 Method of Measurement

NOTE: See the method described in IEEE Std 269.

5.7 ACOUSTIC STABILITY

The Reference Corner Jig (RCJ) used for acoustic stability testing shall consist of 3 planes (2 sides and a base) of a smooth hard surface, that have a square exposed area of 50 cm x 50 cm, combined to form a corner with mutually perpendicular surfaces as shown in Figure 10.

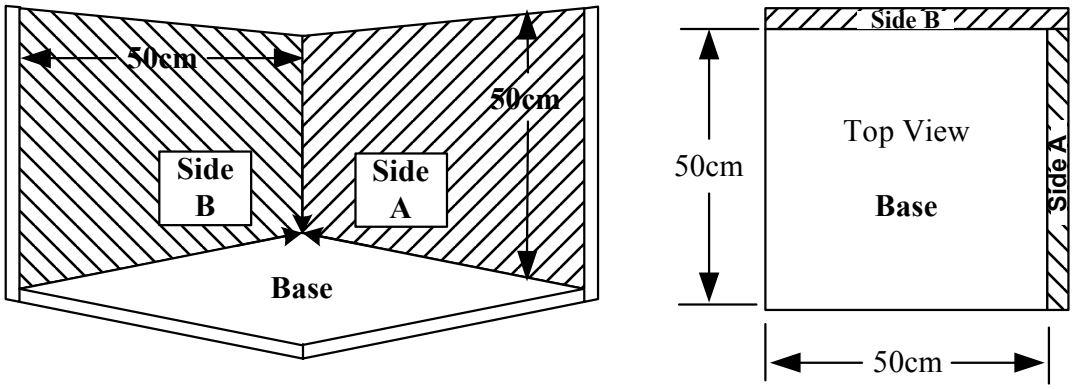


Figure 10 – Reference Corner Jig (RCJ) for stability testing

5.7.1 Requirements

1. When tested as described below, the CPE shall demonstrate acoustic stability by not presenting any signals in the range of 300 Hz to 10 kHz at the SETP greater than –50 dBV when measured with a detector set to maximum hold and having an effective averaging time of 30 to 125 ms.
2. The CPE shall maintain acoustic stability at the Reference and +12 dB Receiver Volume Control Settings when presented to the RCJ, per the orientations listed in Table 12. For volume control settings louder than the +12 dB setting, the CPE should maintain acoustic stability. The CPE is connected to the test circuit in Figure 1 (except no signal shall be applied at the MRP) and tested at 0 km, 2.7 km, and 4.6 km.
3. The CPE shall also maintain acoustic stability for all receiver volume control settings when the handset is removed from, or placed into, the handset cradle. This shall be tested at minimum, reference and maximum volume control settings.
4. The background noise level during acoustic stability testing shall be less than 40 dBA.

750 **Table 12 – Reference corner handset placement for stability test**

Handset Test Orientation	Description
#1	Per Figure 11. Handset placed Face-Down on the Base plane 25cm from the reference corner, equal distant from each side plane. Slowly lift vertically approximately 25cm and then return to the base plane surface.
#2	Per Figure 12. Handset placed On-Side facing parallel to Side plane A, 25cm from the side plane A and equal distance from Side Plane B and the end of the Base plane. Slowly move to the side plane surface, maintaining parallel orientation and then return to the starting point.

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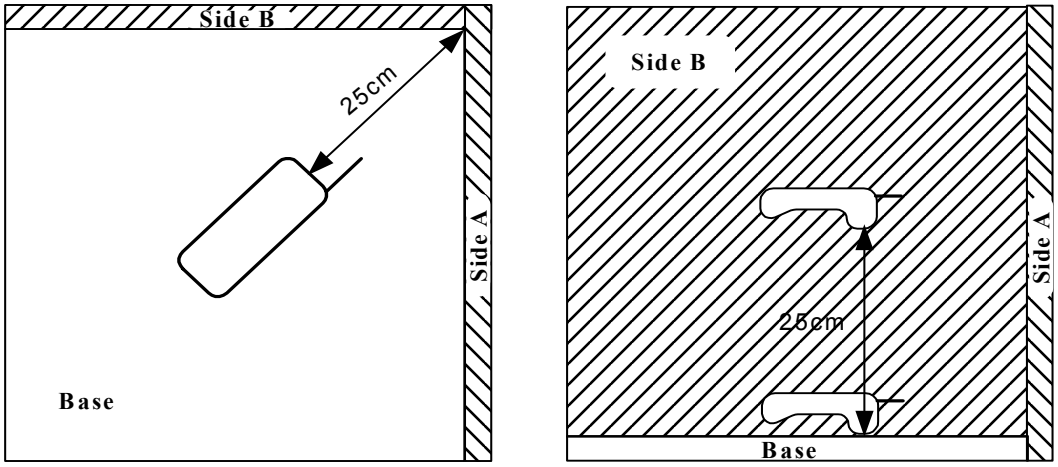

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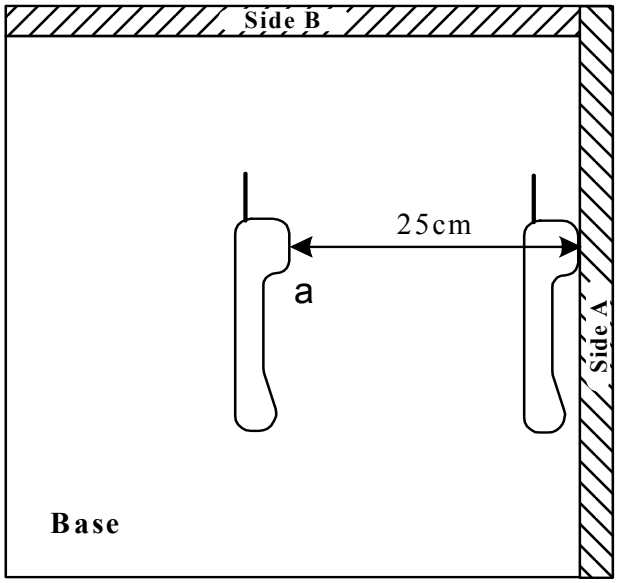
753 **Figure 11 – Stability face down orientation (#1) test**

754



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756 **Figure 12 – Stability on side orientation (#2) test**

Annex A (Normative) – Noise Curves

The send and receive noise weighting curves are shown for reference.

A.1 Send Noise Curves

Psophometric weighting is now being used for send noise measurements in the interest of harmonization with other standards. The psophometric weighting is defined in ITU-T Recommendation O.41 (94). The previously used C-message weighting (defined in IEEE Std 743-1995) is also shown for comparison purposes.

Table A.1 – Psophometric weighting curve

Frequency (Hz)	Relative weight (dB)	Tolerance (± dB)
16.66	-85.0	–
50	-63.0	2
100	-41.0	2
200	-21.0	2
300	-10.6	1
400	-6.3	1
500	-3.6	1
600	-2.0	1
700	-0.9	1
800	–0.0	0 (reference)
900	0.6	1
1000	1.0	1
1200	–0.0	1
1400	-0.9	1
1600	-1.7	1
1800	-2.4	1
2000	-3.0	1
2500	-4.2	1
3000	-5.6	1
3500	-8.5	2
4000	-15.0	3
4500	-25.0	3
5000	-36.0	3
6000	-43.0	–

Table A.2 – C-Message weighting curve

Frequency (Hz)	Relative weight (dB)	Tolerance (± dB)
60	-55.7	2
100	-42.5	2
200	-25.1	2
300	-16.3	2
400	-11.2	1
500	-7.7	1
600	-5.0	1
700	-2.8	1
800	-1.3	1
900	-0.3	1
1000	0.0	0 (reference)
1200	-0.4	1
1300	-0.7	1
1500	-1.2	1
1800	-1.3	1
2000	-1.1	1
2500	-1.1	1
2800	-2.0	1
3000	-3.0	1
3300	-5.1	2
3500	-7.1	2
4000	-14.6	3
4500	-22.3	3
5000	> -28.7	

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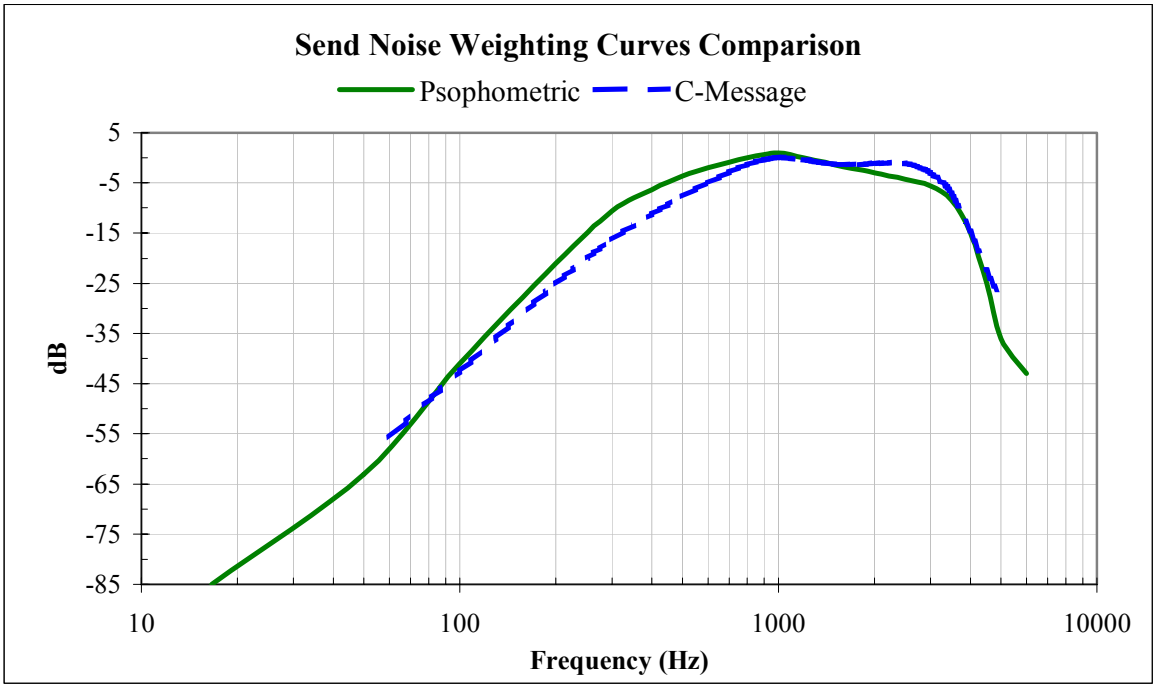

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769 **Figure A.1 – Psophometric and C-Message noise weighting curves comparison**

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A.2 Receive Noise Curve

The A-weighting curve is defined in ANSI S1.4-1983 (Reaff. 2001)

Table A.3 – A-weighting curve

Frequency (Hz)	A-Weighting	Frequency (Hz)	A-Weighting	Frequency (Hz)	A-Weighting
12.5	-63.4	125	-16.1	1250	0.6
16	-56.7	160	-13.4	1600	1.0
20	-50.5	200	-10.9	2000	1.2
25	-44.7	250	-8.6	2500	1.3
31.5	-39.4	315	-6.6	3150	1.2
40	-34.6	400	-4.8	4000	1.0
50	-30.2	500	-3.2	5000	0.5
63	-26.2	630	-1.9	6300	-0.1
80	-22.5	800	-0.8	8000	-1.1
100	-19.1	1000	0.0	10000	-2.5
				12000	-4.3
				16000	-6.6
				20000	-9.3

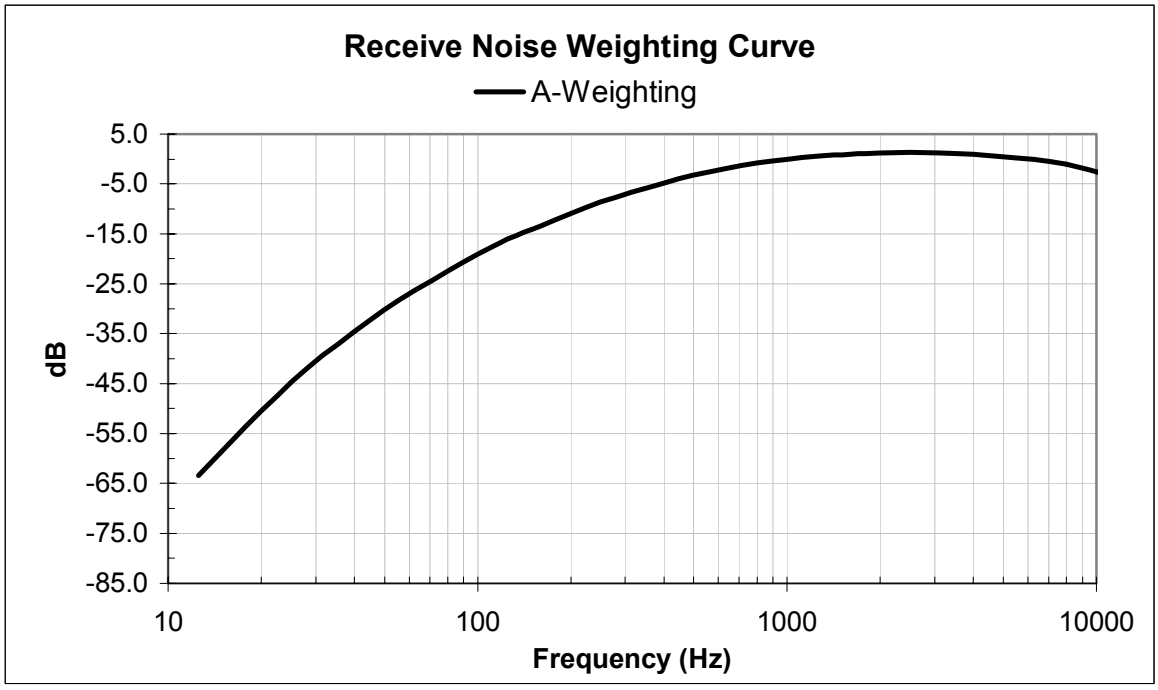

Figure A.2 – A-weighting noise weighting curve

Annex B (Normative) – Calculation of Loudness Ratings

This Annex details the loudness rating calculations and weighting factors relevant to this document. Loudness ratings are used to insure that the loudness of a connection from the Mouth Reference Point (MRP) of the talker to the Ear Reference Point (ERP) of the far end listener is at a satisfactory level. The loudness of the complete path is designated as the Overall Loudness Rating (OLR). The MRP to electrical component is referred to as the Send Loudness Rating (SLR). The electrical to ERP component is the Receive Loudness Rating (RLR). The loudness of the sidetone path of handsets and headsets is referred to as the Sidetone Masking Rating (STMR).

Loudness ratings are used rather than simple level measurements because of better subjective correlation. Loudness ratings more closely account for changes in the ear's sensitivity at different frequencies and its nonlinear response to varying sound levels. The following calculations are based on the 1999 revision of ITU-T Recommendation P.79. Older versions of P.79 should not be used. ITU-T P.79 provides information on the derivation of the loudness rating algorithm.

For use in these loudness ratings formulae, frequency response measurements shall be converted to the R10 format, by using band averaging according to IEEE Std 269.

At each ISO R10 preferred frequency

$$H'(f) = 10\log_{10}\left[\frac{1}{N}\sum_{i=1}^{N} 10^{\frac{H_i}{10}}\right]$$

where

$H'(f)$ = response at the new preferred ISO R10 frequency

f = preferred ISO R10 frequency

N = number of response values within the 1/3 octave band centered at f

i = index for each response value within the 1/3 octave band

H_i = measured response value (in dB)

For the lowest frequency within the band, $i = 1$. For the highest included frequency, $i = N$. The 1/3 octave pass band limit frequencies can be calculated according as:

$$f = 10^{(n/10)\pm 0.05}$$

Where n is the band number.

B.1 Send Loudness Rating (Handset):

$$\mathbf{SLR} = -57.1\log_{10}\sum_{i=\text{Band 4}}^{\text{Band 17}} 10^{(0.1 * 0.175 * (S_{MJ} - W_{si} - K_s))}$$

where:

i Frequency bands from Table 1 of ITU-T P.79 (1999), bands 4-17.

$\boldsymbol{S_{MJ}}$ Send frequency response data (Sensitivity, Mouth-to-Junction) in dB(V/Pa).

$\boldsymbol{W_{si}}$ Send weighting factor from Table 1 of ITU-T P.79 (1999), see Table B.1

$\boldsymbol{K_s}$ Impedance sensitivity correction = 1.76 dB when 900 ohm termination is used in the measurement.

B.2 Receive Loudness Rating (Handset):

$$\mathbf{RLR} = -57.1 \log_{10} \sum_{i=\text{Band 4}}^{\text{Band 17}} 10^{(0.1 * 0.175 * (S_{JE} - W_{ri} - L_E + K_R))}$$

where:

i Frequency bands from Table 1 of ITU-T P.79 (1999), bands 4-17.

S_{JE} Receive frequency response data (Sensitivity, Junction-to-Ear) in dB(Pa/V).

W_{ri} Receive weighting factor from Table 1 of ITU-T P.79 (1999), see Table B.1

L_E Leakage correction is not used when using ear simulators that include leakage such as HATS. (Leakage correction is only used when the handset is sealed to a Type 1 ear. Leakage correction from Table 2 of ITU-T P.79 (1999), see Table B.1

K_R Impedance sensitivity correction = 1.76 dB when 900 ohm termination is used in the measurement.

B.3 Sidetone Masking Rating (Handset):

$$\mathbf{STMR} = -44.4 \log_{10} \sum_{i=\text{Band 1}}^{\text{Band 20}} 10^{(0.1 * 0.225 * (S_{meST} - W_{MSi}))}$$

where:

i Frequency bands from Table 3 of ITU-T P.79 (1999), bands 1-20.

S_{meST} Sidetone frequency response data (Sensitivity, mouth-to-ear) in dB(Pa/Pa).

W_{MSi} Sidetone weighting factor from Table 3 of ITU-T P.79 (1999), see Table B.1.

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847 **Table B.1 – ITU-T P.79 Tables 1, 2 and 3, Weighting factors for calculating loudness ratings**

Band No.	Mid-frequency (Hz)	Send W_{si}	Receive W_{ri}	Receive L_E (dB)	Sidetone W_{MSi}
1	100				110.4
2	125				107.7
3	160				104.6
4	200	76.9	85.0	8.4	98.4
5	250	62.6	74.7	4.9	94.0
6	315	62.0	79.0	1.0	89.8
7	400	44.7	63.7	-0.7	84.8
8	500	53.1	73.5	-2.2	75.5
9	630	48.5	69.1	-2.6	66.0
10	800	47.6	68.0	-3.2	57.1
11	1000	50.1	68.7	-2.3	49.1
12	1250	59.1	75.1	-1.2	50.6
13	1600	56.7	70.4	-0.1	51.0
14	2000	72.2	81.4	3.6	51.9
15	2500	72.6	76.5	7.4	51.3
16	3150	89.2	93.3	6.7	50.6
17	4000	117.0	113.8	8.8	51.0
18	5000				49.7
19	6300				50.0
20	8000				52.8

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Annex C (Normative) – ISO R-40 Frequencies

Table C.1 – ISO R-40, ISO R-10,1/12th Octave and 1/3rd Octave Frequencies

R40 Preferred Frequencies, Hz.	1/12 Oct. Band Center Freq, Hz.	1/3 Oct. Band Center Freq, Hz.	*R10 Preferred Frequencies, Hz.*
90	91.73		
95	97.16	100.00	
100	102.92		***100***
106	109.02		
112	115.48		
118	122.32	125.89	
125	129.57		***125***
132	137.25		
140	145.38		
150	153.99	158.49	
160	163.12		***160***
170	172.78		
180	183.02		
190	193.87	199.53	
200	205.35		***200***
212	217.52		
224	230.41		
236	244.06	251.19	
250	258.52		***250***
265	273.84		
280	290.07		
300	307.26	316.23	
315	325.46		***315***
335	344.75		
355	365.17		
375	386.81	398.11	
400	409.73		***400***
425	434.01		
450	459.73		
475	486.97	501.19	
500	515.82		***500***
530	546.39		
560	578.76		
600	613.06	630.96	
630	649.38		***630***
670	687.86		
710	728.62		
750	771.79	794.33	
800	817.52		***800***
850	865.96		
900	917.28		
950	971.63	1000.00	
1000	1029.20		***1000***
1060	1090.18		
1120			

853 **Table C.2 – ISO R-40, ISO R-10,1/12th Octave and 1/3rd Octave Frequencies (Continued)**

R40 Preferred Frequencies, Hz.	1/12 Oct. Band Center Freq, Hz.	1/3 Oct. Band Center Freq, Hz	***R10 Preferred Frequencies, Hz.***
1120	1154.78		
1180	1223.21	1258.93	***1250***
1250	1295.69		
1320	1372.46		
1400	1453.78		
1500	1539.93	1584.89	***1600***
1600	1631.17		
1700	1727.83		
1800	1830.21		
1900	1938.65	1995.26	***2000***
2000	2053.53		
2120	2175.20		
2240	2304.09		
2360	2440.62	2511.89	***2500***
2500	2585.23		
2650	2738.42		
2800	2900.68		
3000	3072.56	3162.28	***3150***
3150	3254.62		
3350	3447.47		
3550	3651.74		
3750	3868.12	3981.07	***4000***
4000	4097.32		
4250	4340.10		
4500	4597.27		
4750	4869.68	5011.87	***5000***
5000	5158.22		
5300	5463.87		
5600	5787.62		
6000	6130.56	6309.57	***6300***
6300	6493.82		
6700	6878.60		
7100	7286.18		
7500	7717.92	7943.28	***8000***
8000	8175.23		
8500	8659.64		
9000	9172.76		
9500	9716.28	10000.00	***10000***
10000	10292.01		
10600	10901.84		
11200			

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Annex D (Informative) – Receive Acoustic Limiting

Previous versions of this standard have included Receive Acoustic Limiting requirements. These requirements are now covered in UL: 60950.

Annex E (Informative) – Loudness Ratings For Telephones On PBX, VGs and DLCs

The 0 km and 2.7 km normative loudness rating requirements specified in Table 3 for SLR and in Table 8 for RLR for are illustrated in Figure E.1 using solid lines. The ideal telephone specified in this standard has approximately 2 dB of loop compensation, i.e., attenuation of send and receive sensitivities for 0 km loops relative to 2.7 km loops. This is done by sensing the DC current supplied by the 48 volt feed of the DEO (digital end office). The current supplied to the telephone on very short loops is approximately 100 mA, but 2.7 km of cable attenuates the current to approximately 37 mA. In practice many telephones are designed with 2 dB of loop compensation, but in addition some telephones are designed to have even more loop compensation and other telephones are designed without any loop compensation. The dotted lines between 0 km and 2.7 km illustrate a telephone without loop compensation, based on an assumed attenuation rate of 1.5 dB/km for 26 AWG cable or 4 dB total attenuation for 2.7 km of cable.

To complicate matters further, many modern DEOs save money by limiting the current to a maximum of approximately 50 mA, reducing the effect of loop compensation. The purpose of loop compensation is to reduce the loudness rating variation with respect to loop length, which improves the user experience. For instance, the loudness ratings of digital telephones are independent of line length, so they provide consistent audio performance regardless of distance.

Modern PBXs and VGs (voice gateways) also limit the current to analog sets, but they limit at a much lower current, around 28 mA to reduce the size and expense of the power supplies. This level of current is similar to what a telephone would normally obtain from a DEO 4.6 km away, yet telephones connected to PBXs and VGs are generally on very short loops. To put it another way, telephones behind PBXs and VGs are often on very short AC loops and very long DC loops. This eliminates any "normal" loop compensation that might be designed into a telephone. Of course, there are more sophisticated analog telephones with "abnormal" loop compensation that can deal with this condition, but they are the minority. Most analog telephones connected to PBXs and VGs behave as if they are on 4.6 km loops, i.e., no loop compensation, without the 7 dB of AC attenuation. The resulting loudness ratings are RLR = -7 dB, the pointed labeled ONS RLR, and SLR = 4 dB, the point labeled ONS SLR. ONS stands for On Premise Station, a dated term referring to telephones in the same building as the PBX. For PBX and VG loss planning purposes, ONS RLR = - 7 dB and ONS SLR = 4 dB are the reference loudness rating values not only for Enterprise equipment, but cable and DSL VGs in residential and SOHO situations. According to ITU-T Recommendation G.100.1, Clause 6.3, SLR > 2 dB gives a reasonable protection against objectionable speech clipping.

In summary at 0 km, loop compensated telephones are specified to have RLR = -5 dB and SLR = 6 dB, while telephones without loop compensation and most telephones in ONS applications are specified to RLR = -7 dB and SLR = 4 dB. It is worth noting that the termination impedance of DEOs is 900 ohms, but the termination impedance of PBXs and VGs is 600 ohms.

For DEO loss planning purposes, the loudness ratings specified at 2.7 km, RLR = - 3 dB and SLR = 8 dB, are the reference loudness rating values. The loss planning acronym for this these values is AAL (Analog Access Line). The AAL objectives originate in an ITU-T Recommendation and are illustrated in Figure F.1. Not all telephones are connected to PBXs and VGs with short loops. For PBX and VG loss planning purposes, there is an additional condition called OPS or Off Premise Station, that also specifies RLR = - 3 dB and SLR = 8 dB at 2.7 km for the reference loudness rating values. The only difference between AAL and OPS is the 900 ohm vs. 600 ohm termination.

The attenuation between 2.7 km and 4.6 km is assumed to be 3 dB, based on the attenuation rate above. However, Table 3 and Table 8 specify only a 2 dB difference in loudness ratings. This implies 1 dB of loop compensation between 2.7 km and 4.6 km, which reflects the objectives of the past to reduce loudness variation, not the current realities of equipment working against loop compensation. Figure E.1, simply illustrates 3 dB of loss between 2.7 km and 4.6 km.

Not all DEOs employ 48 volt DC feeds with a nominal loop length of 2.7 km. Digital Loop Carrier (DLC) is now common place in the network. DLCs typically use low constant current feeds similar to PBXs and VGs and they typically have shorter nominal loops than 2.7 km. For the purposes of this revision of this standard, it is assumed that the loudness ratings of CPE connected to DLCs are between the ONS values for PBXs/VGs and the 0 km loudness ratings specified in Table 3 and Table 8, however this situation requires further study. Loudness ratings for telephones on a DLC may be a requirement in a future version of this standard.

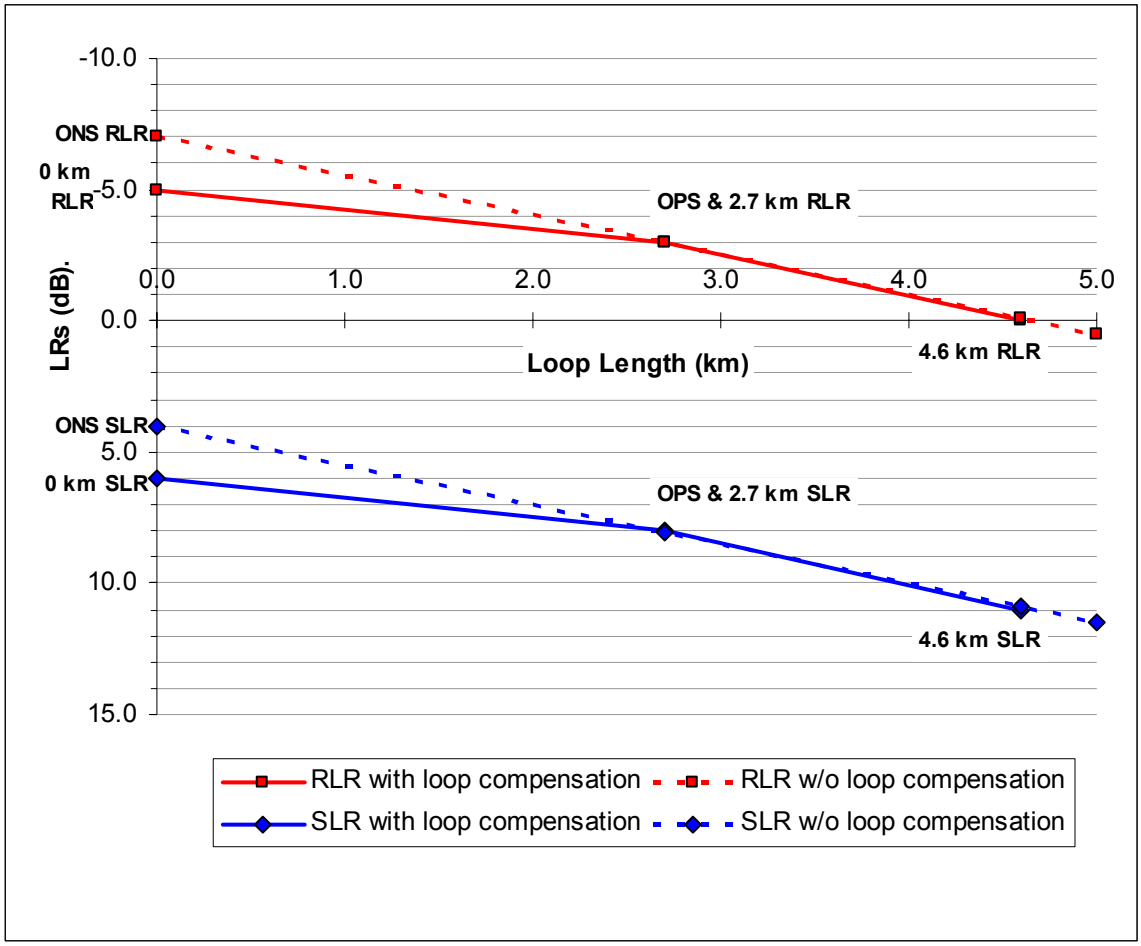
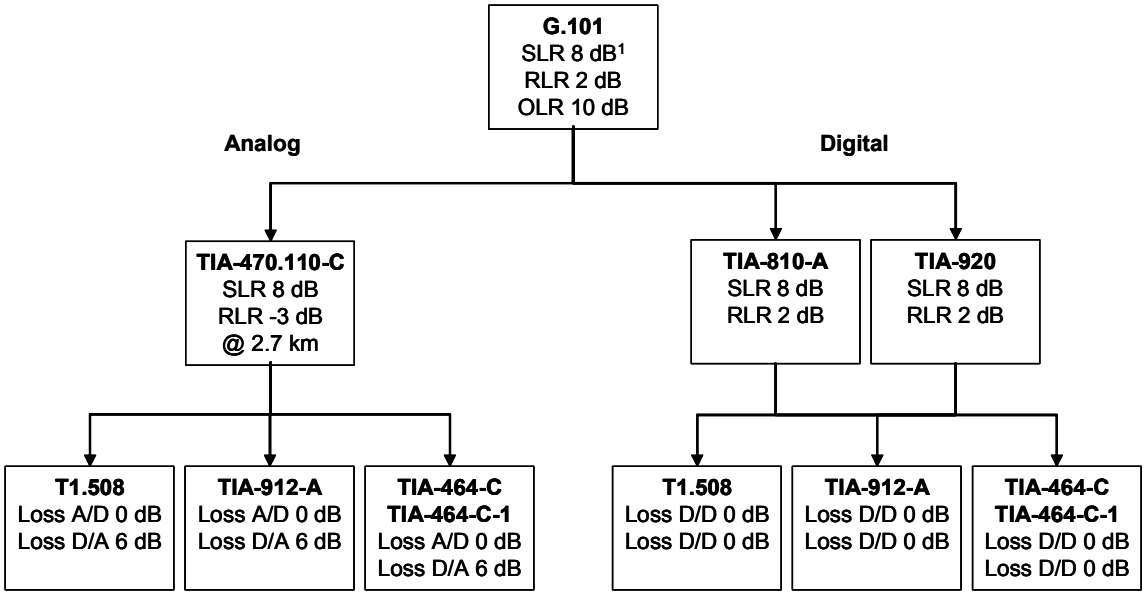


Figure E.1 – Loudness Rating Relationships for PBXs, VGs and DEOs

Annex F (Informative) – Loss Plan and Loudness Rating References and Hierarchy



ESLR = SLR + Loss A/D = 8 + 0 = 8 dB
ERLR = RLR + Loss D/A = -3 + 6 = 3 dB
OLR = SLR + Loss A/D + Loss D/D + Loss D/A + RLR
= 8 + 0 + 0 + 6 - 3 = 11 dB

ESLR = SLR + Loss D/D = 8 + 0 = 8 dB
ERLR = RLR + Loss D/D = 2 + 0 = 2 dB
OLR = SLR + Loss D/D + RLR = 8 + 0 + 2 = 10 dB

Where: ESLR and ERLR are the "Equivalent" SLR and RLR at the 0 dBr point.
Note 1: According to ITU-T Recommendation G.100.1, Clause 6.3, SLR > 2 dB gives a reasonable protection against objectionable speech clipping.

Figure F.1 – Hierarchical Relationship between Loudness Rating and Loss Plan Standards

Annex G (Informative) – Nominal Frequency Response Characteristics and Loudness Rating Calculations

This standard specifies the send and receive loudness ratings, as well as the sidetone loudness, using ITU-T Recommendation P.79. Previous revisions of this standard used the methodology of IEEE Std 661. The intent of this annex is to provide information on the approximate relationship between the loudness ratings obtained using the two methods.

In order to accomplish this, nearly ideal send and receive response curves were drawn through the specified frequency response templates. The overall levels of the curves were adjusted to produce the specified nominal loudness rating values of SLR = 8 dB and RLR = 2 dB. The corresponding IEEE Std 661 objective loudness rating parameters, TOLR = -49.4 dB and ROLR = 52.7 dB, were then calculated from the same frequency response curves, taking into account the correction from 600 ohm to 900 ohm impedances.

For sidetone, the send and receive frequency response curves were added together and an appropriate overall level shift was made to provide the nominal sidetone masking rating value of STMR = 15 dB. This simulates a design in which a part of the send signal is fed to the receive path via a frequency independent loss path to produce the sidetone signal. The corresponding IEEE Std 661 sidetone loudness rating parameter, SOLR = 5.5 dB, was then calculated from the response curve.

Figure G.1and Figure G.2 show the send and receive sensitivity curves, and Table G.1 provides the details of the various loudness rating calculations. The first column of the table lists the ISO preferred 1/3 octave frequencies from 100 to 8000 Hz; it also includes the frequencies 300 Hz and 3300 Hz that are the endpoints of the IEEE loudness rating calculation bandwidth. The next three columns of the table list the nominal send, receive, and sidetone sensitivities at each of the specified frequencies.

There are then seven columns in the table associated with the P.79 loudness rating calculations. The first three of these columns provide the W-weights for send, receive and masked sidetone used in the P.79 calculation algorithm. The fourth column is the L_E factor included in the receive calculation to take account of acoustic leakage between the ear and the handset. Finally, there are three columns providing the SLR, RLR, and STMR calculations. The table entries at each frequency indicate the relative weight that frequency has on the overall loudness, taking into account the frequency response, the W-weight, and (for receive) the L_E factor. The send and receive loudness ratings are calculated over the frequency range from 200 to 4000 Hz, whereas the sidetone masking rating is calculated over the range from 100 to 8000 Hz, both in keeping with the latest version of P.79. The actual loudness rating is shown in the last row of the table.

The final three columns of the table are associated with the IEEE Std 661 loudness ratings. Since this method uses a flat weighting in which equally spaced intervals on a logarithmic frequency scale receive equal weight, no separate column of weights is necessary. Neither is any The final three columns of the table are associated with the IEEE Std 661 loudness ratings. Since this method uses a flat weighting in which equally spaced intervals on a logarithmic frequency scale receive equal weight, no separate column of weights is necessary. Neither is any allowance made for acoustic leakage between the ear and the handset in the receive calculation. The table entries in each column represent the relative contribution made by the frequency interval from the frequency just above the entry in the table to the frequency of the entry in the table. The actual loudness ratings again appear in the last row of the table.

The exact calculations in the table give the following relationships between P.79 and IEEE Std 661 loudness ratings:

SLR (P.79) = TOLR (IEEE) + 57.4 dB

RLR (P.79) = ROLR (IEEE) - 50.7 dB

STMR (P.79) = SOLR (IEEE) + 9.5 dB

These relationships take into account a nominal 600 ohm impedance for the P.79 calculations and a nominal 900 ohm impedance for the IEEE Std 661 calculations. There are no other known sources of systematic difference between the two methods, other than the effects of the weighting functions on various frequency response shapes, for the receive calculations. For the send and sidetone measurements, the method that has previously been used to calibrate the artificial mouth† would result in actual measured sensitivities and transmit levels that are about 0.5 dB lower than those calculated in Table G.1.

Based on the above, the following approximate relationships between P.79 and IEEE Std 661 loudness ratings can be established:††

SLR (P.79) = TOLR (IEEE) + 57 dB

RLR (P.79) = ROLR (IEEE) - 51 dB

STMR (P.79) = SOLR (IEEE) + 9 dB

The conversion factors for real handsets may vary from the above idealized conversion factors due to mechanical shape, transducer sensitivity and transducer frequency response. Therefore, the actual conversion factors for each handset model must be determined by measurement.

† The 1979 version of IEEE Std 661 used a free field microphone aligned coaxially with the artificial mouth for its calibration. The revised 1992 version of IEEE Std 269 adopts the ITU method of using a pressure microphone aligned perpendicular to the mouth axis. The latter method very nearly approximates a true free field measurement, whereas the former method indicates a sound pressure that is about 0.5 dB too high. This results in lowering the drive level to the artificial mouth by a corresponding amount.

†† ITU-T Supplement No. 19 to the P-Series Recommendations lists slightly different values for these relationships. In particular, it uses +56 for the send relationship, -50 for receive, and +8 for sidetone.

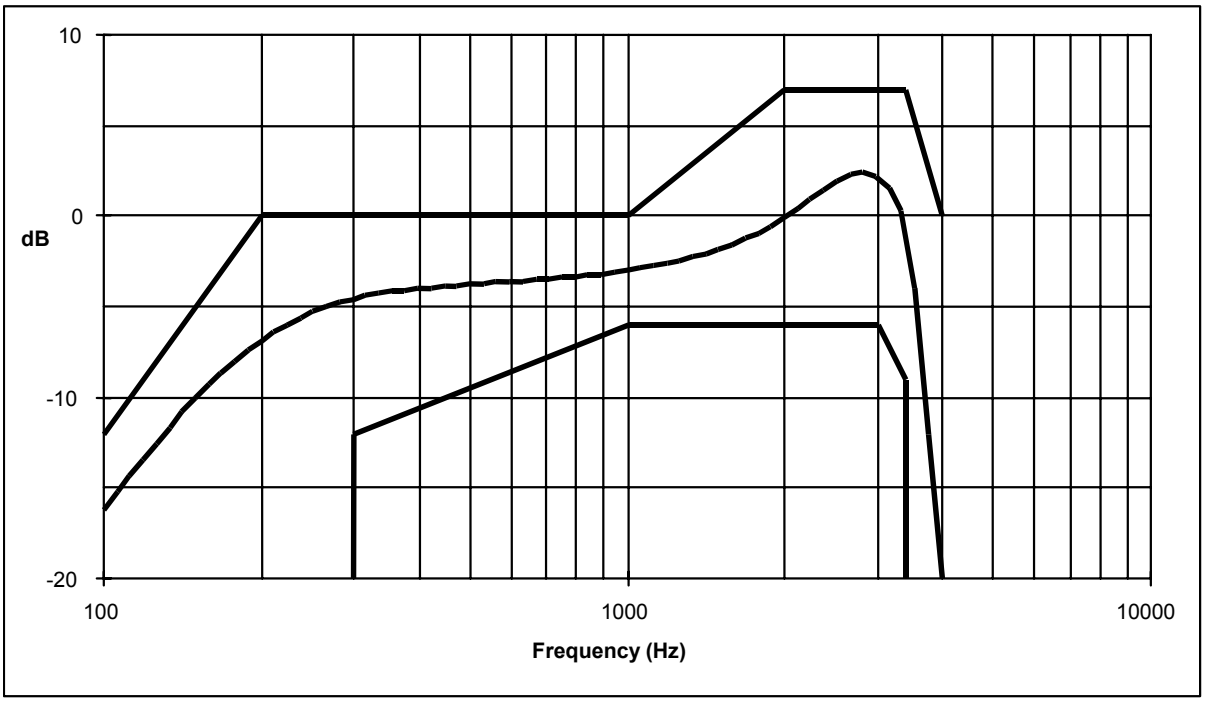


Figure G.1 – Nominal send frequency response

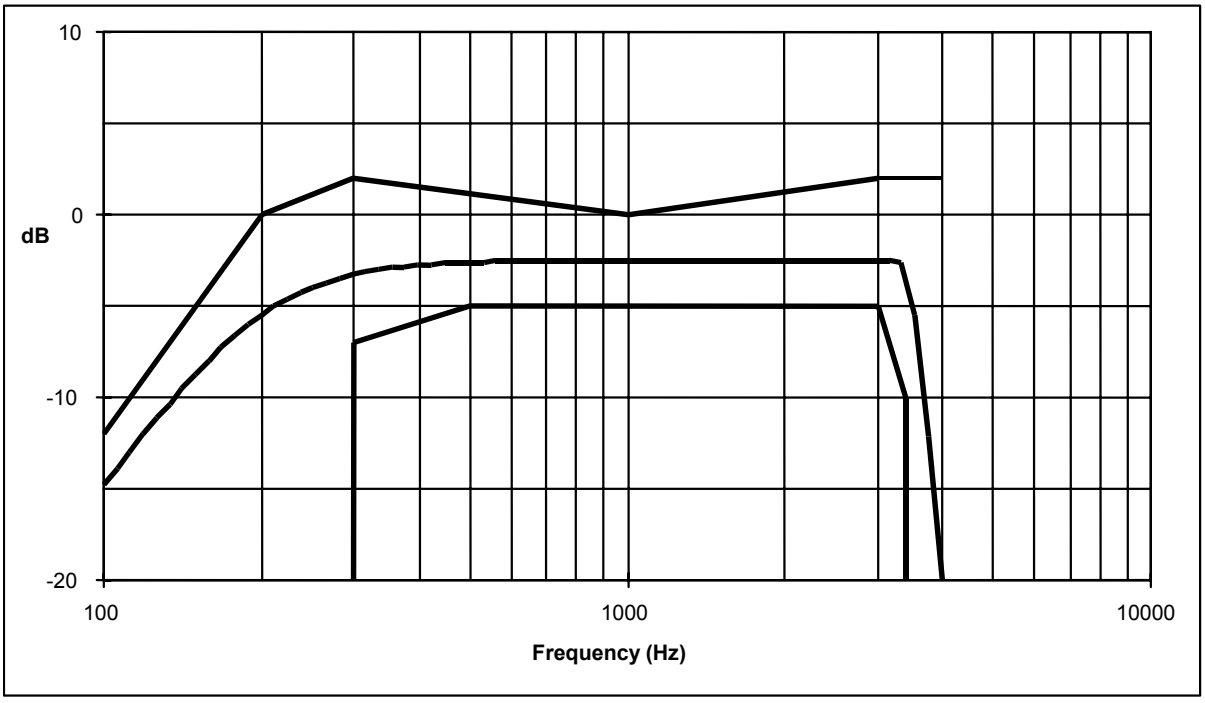


Figure G.2 – Nominal receive frequency response

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1017 **Table G.1 – Nominal Frequency Response Characteristics and Loudness Ratings Calculations**

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	Nominal	Nominal	Nominal	ITU-T P.79							IEEE 661		
Frequency Hz	Send dBV/Pa	Receive dBPa/V	Sidetone dB	Ws	Wr	Wms	LE	SLR	RLR	STMR	TOLR	ROLR	SOLR
100	-26.54	-3.18	-31.82			110.4				0.0006			
125	-23.01	0.35	-24.77			107.7				0.0010			
160	-19.59	3.76	-17.94			104.6				0.0017			
200	-17.22	6.12	-13.21	76.9	85.0	98.4	8.4	0.0225	0.0296	0.0031			
250	-15.68	7.64	-10.15	62.6	74.7	94.0	4.9	0.0426	0.0549	0.0045			
300	-14.95	8.35	-8.71										
315	-14.81	8.47	-8.45	62.0	79.0	89.8	1.0	0.0452	0.0559	0.0062	0.0097	0.0329	0.0135
400	-14.37	8.86	-7.61	44.7	63.7	84.8	-0.7	0.0924	0.1127	0.0083	0.0484	0.1633	0.0682
500	-14.14	9.02	-7.23	53.1	73.5	75.5	-2.2	0.0664	0.0811	0.0138	0.0460	0.1547	0.0657
630	-13.95	9.08	-6.97	48.5	69.1	66.0	-2.6	0.0806	0.0987	0.0228	0.0481	0.1612	0.0692
800	-13.71	9.11	-6.71	47.6	68.0	57.1	-3.2	0.0844	0.1059	0.0367	0.0503	0.1670	0.0725
1000	-13.38	9.12	-6.37	50.1	68.7	49.1	-2.3	0.0773	0.0993	0.0565	0.0477	0.1562	0.0688
1250	-12.86	9.13	-5.84	59.1	75.1	50.6	-1.2	0.0549	0.0734	0.0537	0.0488	0.1562	0.0704
1600	-11.90	9.13	-4.88	56.7	70.4	51.0	-0.1	0.0629	0.0849	0.0553	0.0561	0.1728	0.0810
2000	-10.47	9.13	-3.45	72.2	81.4	51.9	3.6	0.0357	0.0469	0.0568	0.0540	0.1562	0.0780
2500	-8.51	9.13	-1.49	72.6	76.5	51.3	7.4	0.0380	0.0490	0.0649	0.0591	0.1562	0.0853
3150	-8.81	9.13	-1.79	89.2	93.3	50.6	6.7	0.0192	0.0256	0.0663	0.0638	0.1618	0.0921
3300	-9.64	9.13	-2.62								0.0125	0.0326	0.0180
4000	-34.45	-10.87	-47.43	117.0	113.8	51.0	8.8	0.0022	0.0046	0.0061			
5000	-60.10	-30.87	-93.08			49.7				0.0006			
6300	-65.29	-30.87	-98.27			50.0				0.0005			
8000	-70.17	-30.87	-103.15			52.8				0.0003			
								SLR	***RLR***	***STMR***	***TOLR***	***ROLR***	***SOLR***
								8.0	***2.0***	***15.0***	***-49.4***	***52.7***	***5.5***

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Annex H (Informative) – Use Of The Free Field As The Telephonometric Reference Point

Current performance requirements are based on measurements referred to the ERP. Future requirements may be based on measurements referred to the free field. This annex provides background information on this concept.

One goal of a telephonic experience is to simulate a conversation where two people are one meter apart, talking to each other. Now insert a complete telephone system between our two talkers. In a perfect world, the quality of the conversation would be the same with a telephone system and in free space. This is called the orthotelephonic reference.

Consider a loudspeaker with a perfectly flat free field frequency response through the audio band (Figure H.1):

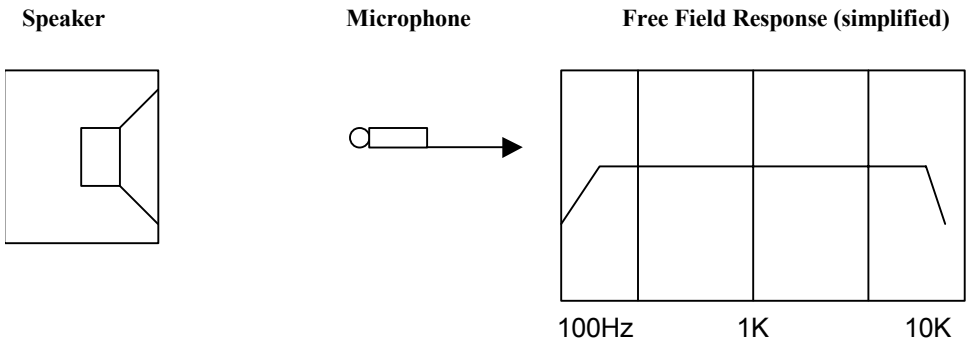

Figure H.1 – Flat free field loudspeaker response

Play the same speaker into a HATS ear simulator, and the result is a 17 dB peak at 2.8 kHz. (For more complete data, see ITU-T P.57 and P.58) (Figure H.2).

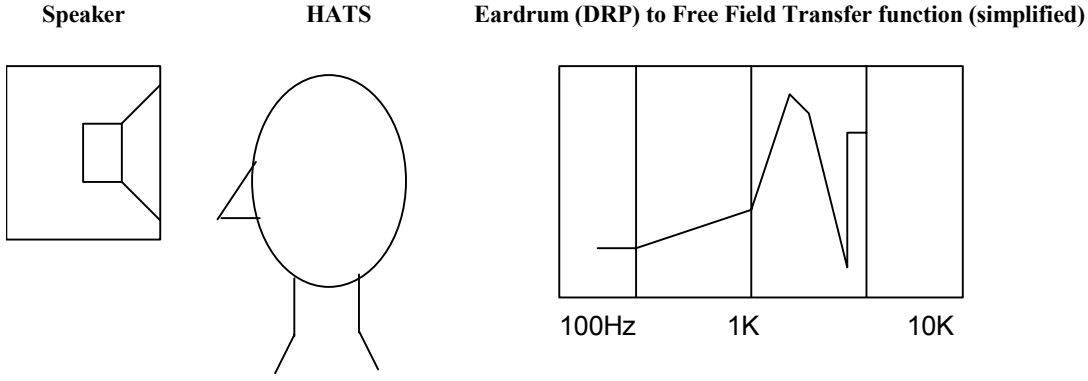

Figure H.2 – HATS measurement of same loudspeaker

The HATS ear simulator replicates the resonances that occur in a typical human pinna and ear canal system, and measures at the (ear) Drum Reference Point or DRP. It is because of the pinna and the resonances in the ear canal that a loudspeaker with a flat free field response will not measure flat into a HATS, at the DRP. Therefore, if a telephone receiver or headset is to sound the same as a hypothetical flat speaker in the free field, the frequency response at the DRP should follow the free field curve(s) referenced in ITU-T P.57 and P.58. (Figure H.3)

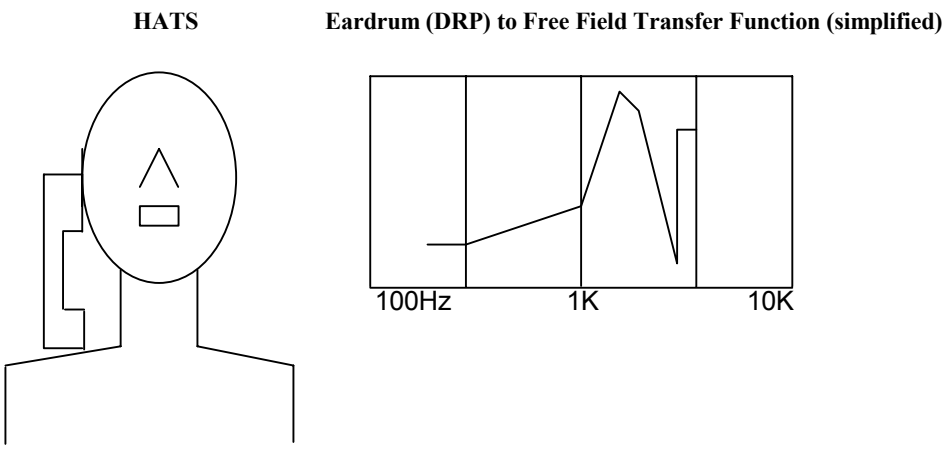

Figure H.3 – HATS measurement of handset telephone

Most telephone manufacturers are more familiar with the Type 1 ear simulator. This type of simulator uses the Ear Reference Point (ERP) rather than the DRP, which results in a different frequency curve shape. Using the above hypothetical receiver tested into a Type 1 ear simulator yields a frequency response that looks like Figure H.4:

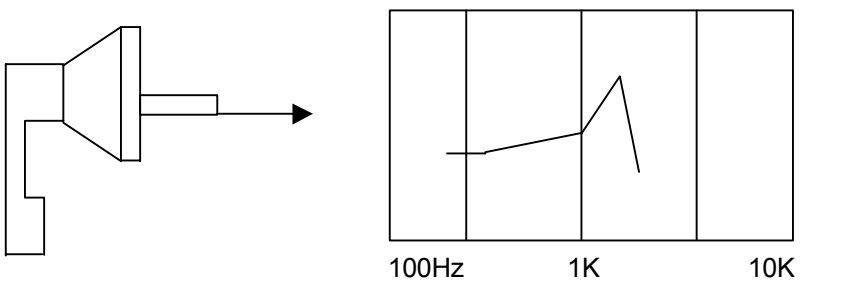

Figure H.4 – Type 1 ear simulator measurement of handset telephone

The important thing to remember is that the above curves, using either Type 1 simulators or HATS, all can be referenced to a free field response. Another way of looking at it is that if you want your handset or headset to sound like a flat loudspeaker in the free field, e.g. simulating the orthotelephonic reference, the frequency response should look like either the ERP or DRP to free field transfer function curves above.

The complete orthotelephonic response is due to the combination of frequency responses in the send, network, line and receive paths of an overall (end-to-end) connection. The exact distribution of frequency response shaping in these paths is outside the scope of this annex.